BAVC Intern Training Workbook

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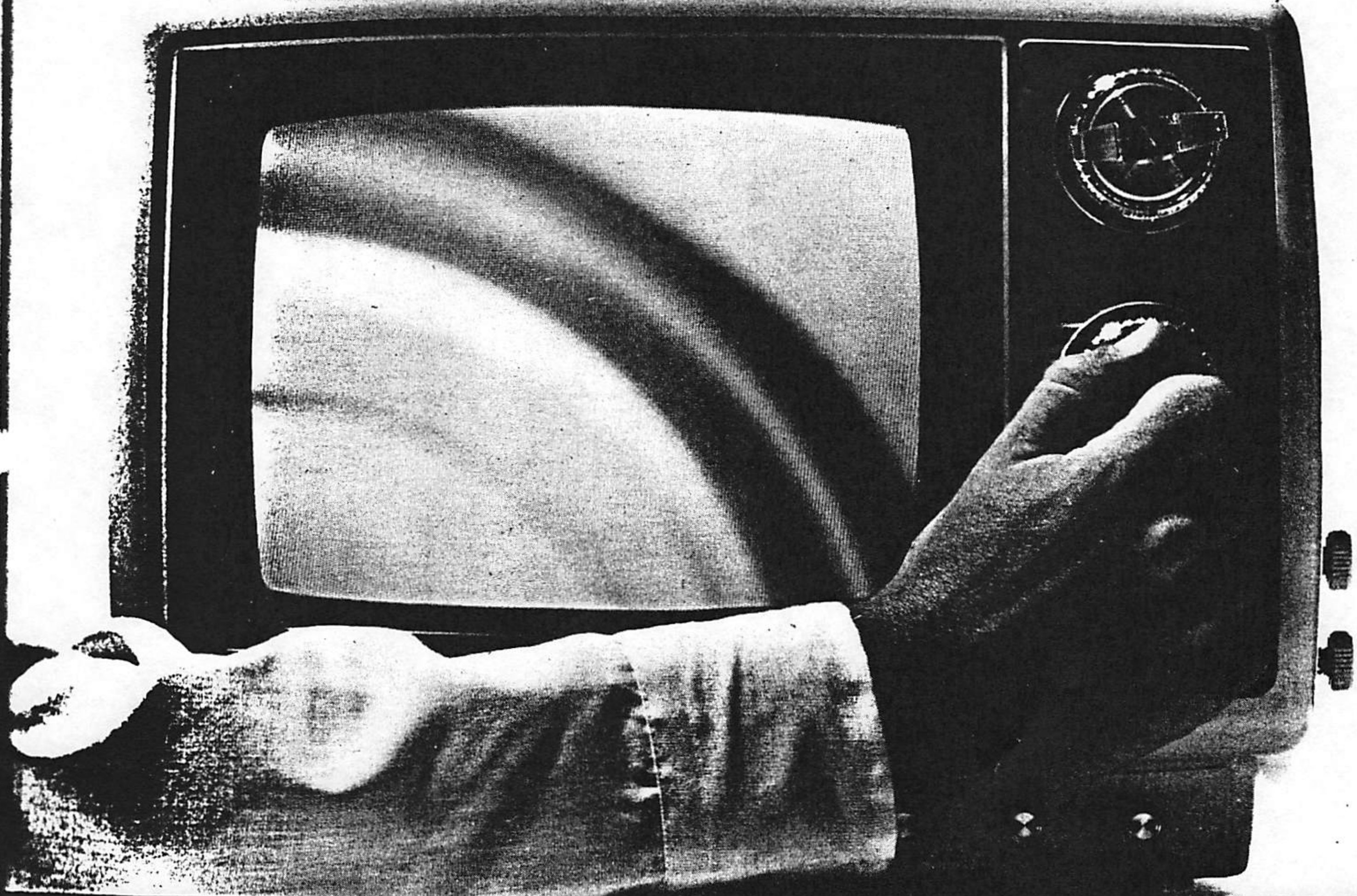
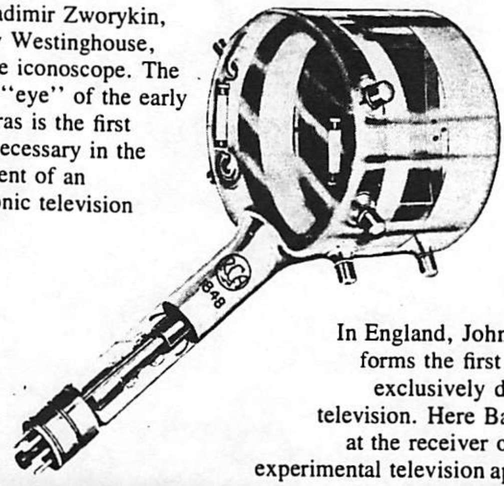
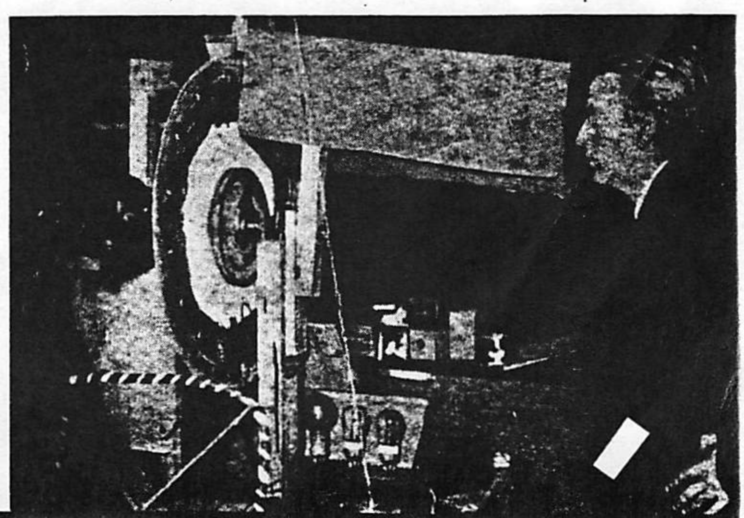
I. THE REVOLUTIONARY MACHINE

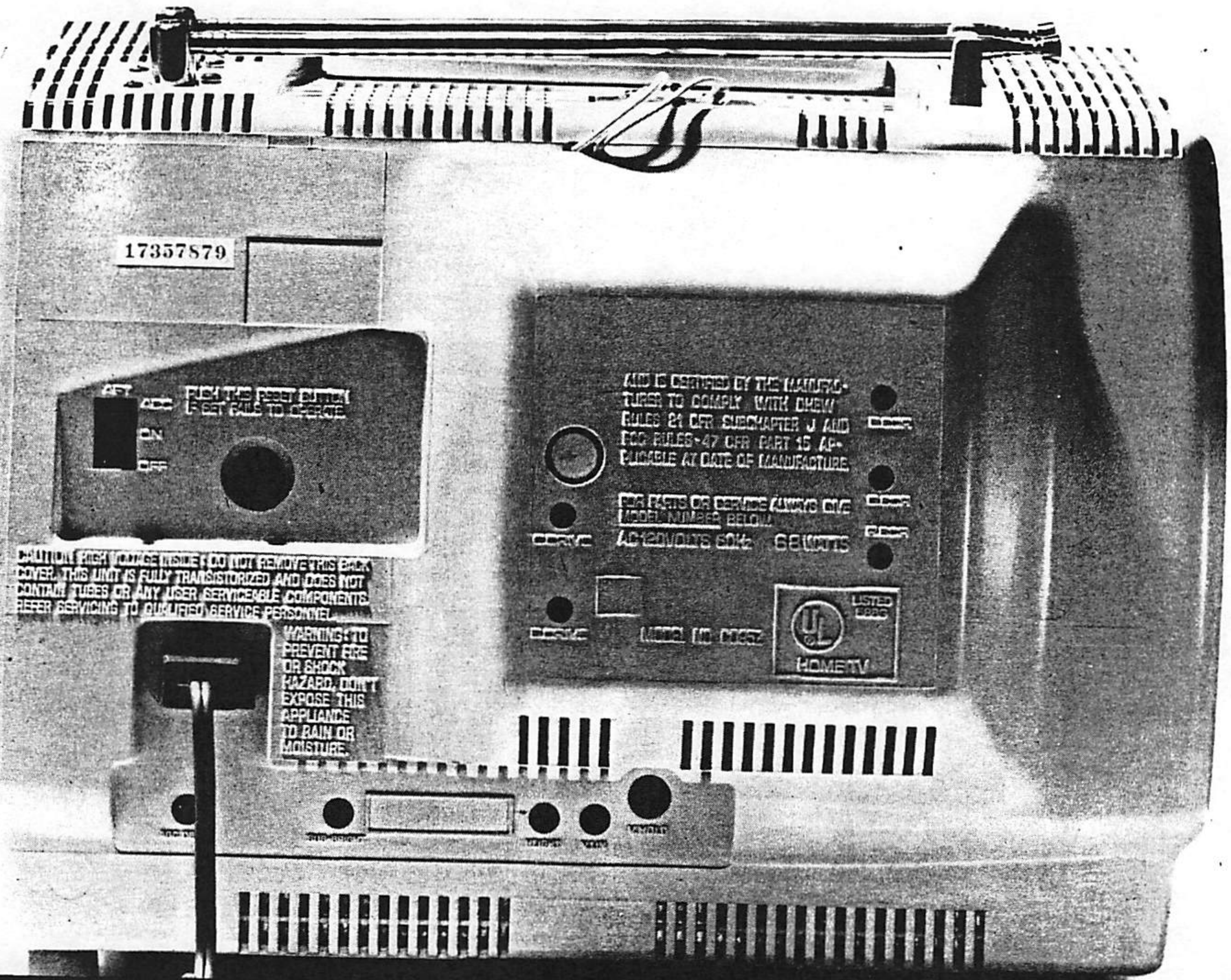


1923. Vladimir Zworykin, funded by Westinghouse, invents the iconoscope. The photocell "eye" of the early TV cameras is the first element necessary in the development of an all-electronic television system.

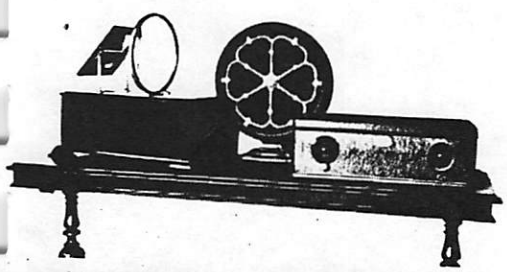




In England, John L. Baird forms the first company exclusively devoted to television. Here Baird looks at the receiver of his first experimental television apparatus.

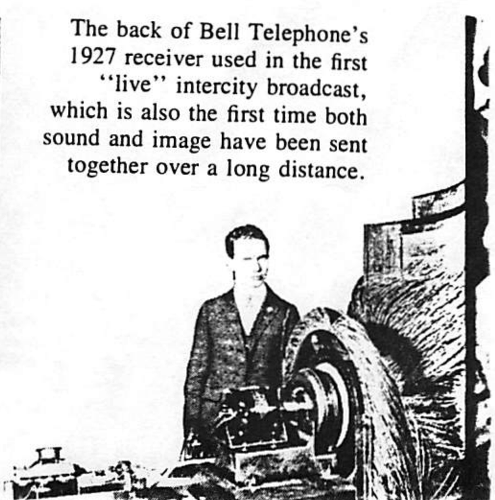


An early radiovisor in which the image is caught from a hole in the box by the mirror and reflected through a magnifying glass. Its inventor, C. Francis Jenkins, transmits a mechanically scanned photo of President Harding from Washington to Philadelphia. In 1925, Jenkins will be the first to transmit the image of a moving object.



1925. Baird makes the first demonstration of wireless television. In 1929, he will begin daily transmissions over the BBC.

The back of Bell Telephone's 1927 receiver used in the first "live" intercity broadcast, which is also the first time both sound and image have been sent together over a long distance.



LIGHT

A black and white image only makes use of one characteristic of light. Which one is it?

A car that is described as rich red as opposed to pale red is describing which characteristic?

A blue book versus a green book versus a yellow book is defined by which characteristic of light?

A white object has how much saturation?

A black object has how much brightness?

A white object has *which* hue?

A black object has *which* hue?

A black object has how much saturation?

The first thing light encounters in a video system is the ______________?

A lens serves what purpose?

What would be missing in a video or vision system without a lens?

How does a mirror affect light?

Modern light theories state that light behaves how?

Examples of light emitting objects are:

Examples of reflected light are:

An object that is photoemissive is ____________________________.

Describe the Additive Color Theory:

Sending white light through a prism will produce what at the other end?

Combining all the colors of the rainbow into a prism will produce what?

What are the three primary colors in modern video systems?

What are the three characteristics of light that the eye can differentiate?

In Additive Color Theory, Yellow is a combination of _______________________.

Magenta is a _____________________________.

Cyan is a _______________________________.

Black is ____.

White is ____.

A filter does what?

A Yellow filter passes what color frequency of light?

A Yellow filter absorbs which color frequencies of light?

Electromagnetic Radiation

Define frequency:

What is a Hertz?

Define the following prefixes:

kilo-
Meager-
Giga-
Tera-

centi-
milli-
micro-
nano-
pico-

X-Rays, light, microwaves, gamma rays, and radio waves are all part of ____________.

What type of radiation is at the "slow" end of the frequency spectrum?

What type of radiation is at the high-frequency end of the spectrum?

Which color has a higher frequency, red or violet?

Which color frequency is just below the range of visible light?

What medium does sound travel in?

What medium does light travel in?

What is the frequency response (range) of human hearing

Name the main color distinctions of the rainbow:

What three characteristics distinguish one color from another?

What determines the color of an object?

Why can't we see X-Rays?

Why don't humans hear radio waves *before* they are picked up by a radio?

Radio Frequencies and Modulation

What is the carrier of information in radio / television broadcast?

What is the difference between light, microwaves, X-rays and radio waves?

What do X-rays, radio waves, light waves and microwaves have in common?

How can radio waves be used to transmit information?

What is modulation?

Describe amplitude modulation:

Describe frequency modulation:

What is the range of carrier frequencies used in amplitude modulation?

What is the FM range of carrier frequencies?

What is the frequency range of light radiation?

Electricity

List some examples of electricity that occur in nature:

All matter is made up of ______________.

Name the subatomic particles:

Specify their charges:

What causes electricity?

What does electrical potential mean?

What is another name for electrical potential?

What is electrical current?

What is another name for electrical current?

A strong voltage will create a strong or weak current?

A weak current implies a strong or weak voltage?

If there is a high current flowing through a wire, what does this mean about the two terminals (or ends) of the wire?

What blocks electrical current from flowing efficiently?

What causes electrical current to be diminished in a material?

What is an electrical conductor?

What is an electrical insulator?

What is the unit of measurement for electrical resistance?

What is Ohms's Law?

What does Ohm's Law convey?

Do batteries produce alternating current or direct current?

What is the difference between alternating current and direct current?

What are the advantages of alternating current?

What are advantages of direct current?

What are the AC specifications (frequency, peak voltages) in the United States?

What is the British AC standard?

What is a name for resistance to an alternating current?

What are possible means of getting power to a video camera?

Transduction

What is transduction?

Name several types of audio transducers:

Describe several forms of visual transducers:

What is useful about transduction?

In the case of audio and video technology, transducing energy into electrical energy is only for processing. What must happen to the signal for final listening or viewing?

Time

Name some ways to detect the passage of time:

Why does a bicycle wheel or a horse's legs tend to blur when moving quickly?

The time it takes for the earth to orbit the sun is called?

The time it takes for the moon to orbit the earth is called ______________.

A day is a measurement of what cycle?

How many seconds are in a year?

How many picoseconds are in one second?

Do things happen in less than a second?

Do things happen in less than a trillionth of a second?

How many lines are in an NTSC video frame?

How many NTSC video frames are in a second (approximately)?

How many lines per second in an NTSC video signal?

What is the frequency of lines (per second) of NTSC video?

Could video lines be considered a measurement of time?

How long does it take to draw one line of NTSC video?

How many NTSC video lines are in a day?

Video

What is the purpose of using video?

What is video?

What are some advantages of video (over film)?

A video camera attempts to mimic what natural phenomenon?

What does transduction mean?

What is a photoelectric material?

How does a video tube (transducer) convert light into electricity?

How is an entire image (many points of light) transduced to an electrical signal that can only represent one luminance level at a time?

How quickly can human eyes detect change?

What is the purpose of scanning with an electron gun?

How is the electron beam aimed?

Why does the electron gun shut off after scanning a line of video?

What is horizontal and vertical blanking?

Scanning first the odd lines and then the even lines is called what?

How many electron guns are in a color video camera or monitor?

Color Video Signals

p.100 What determines the color of an object? the human eyes interpretation of frequen-ies of electromagnetic spectrum

Light is part of the electromagnetic spectrum.

What is the additive color theory of light?
color is produced by the combination of primary colors. White generated by 3 primary colors @ full strength.

What are the three primary colors in video?
Red Green + Blue

What is the transducer in a video camera?
"cathode ray tube" [CRT]

What does the video camera transducer do?
turns light into electrical energy/signals

What does a video monitor transducer do?
turn electrical energy/signals into light

What device produces the scanning behavior of video signals?

What is the purpose of scanning?

Why aren't Red, Green and Blue signals broadcast?
better to broadcast them whole as ONE signal not to risk losing one color in the image.

In what ways are black-and-white and color TV signals compatible?

What wavelength (or color) of light do humans perceive the highest detail?

Colorimetery is the science of ____________________________.

What is the RGB to luminance (Y) equation?

Which of the three signals R, G, and B does a black-and-white television use?

What are the two color difference channels called?

Name the three color component signals derived from R, G, B:

With the signal Y (luminance) and R-Y, how can R be derived?

From Y, R and B, how is G derived?

How is Y/C derived from Y, R-Y and B-Y?

How is NTSC composite derived from Y/C?

Processing RGB signals to NTSC composite is known as ______________________.

Deriving RGB from the NTSC composite signal is known as ___________________.

A color video monitor has how many electron guns?

Video - the Monitor

What is the name of the transducer in a video monitor?

What does the video monitor transducer do?

What allows a camera and a monitor to scan their electron guns in time with one another?

What are some problems if the camera and monitor electron guns are not scanning together, in sync?

Dubbing

BSP In = YRB OUT - YRB, Composite

What type of video input/output does a BetacamSP deck have?

What video input/output does a Hi-8 deck have? H8 = Y/C

Name two types of video input/output for 3/4" U-matic decks: Composite

What is the CTDM signal used for?

What devices use component serial digital video connections?

What kind of connector is used for professional audio? XLR or Balanced

What kind of audio connectors are used on consumer VHS decks? RCA

What is the AES/EBU audio standard for?

What is the name of the consumer digital audio standard?

Name several types of remote connectors:

Name two remote device communications protocols used between decks and edit controllers:

What are the typical tracks on a video tape?

What is the control track for on a tape?

From Hi-8 to BetacamSP, what transcoding of signals would yield the highest quality?

Dubbing 3/4" to VHS would require what kind of video connection?

Most modern decks use which number of pins on remote cables?

What is the current protocol for deck communications?

To connect consumer VTR audio to professional audio means what three things are necessary?

NTSC Video

NTSC stands for ____________________________.

What is the purpose of the NTSC television standard?

Define the following NTSC specifications:

Fields per frame:
Frames per second:
Fields per second:
Lines per field:
Lines per frame:
Voltage of signal (peak to peak):
IRE units of signal (peak to peak):
AMPLITUDE of sync pulse:
AMPLITUDE of color burst:
Blanking level:
Black level:
Peak white level:
Color subcarrier frequency:

Horizontal synchronization pulses occur how often?

Vertical synchronization pulses occur how often?

Where in the NTSC video signal is the color burst located?

Is NTSC a composite signal or a component signal?

What is the difference between a composite and a component signal?

What are some disadvantages to NTSC?

What's the advantage of a composite signal?

How is NTSC video encoded from the original R, G, B color signals from the video camera?

Video Tape Recorders

When was the video tape recorder invented?

What makes recording a video signal on magnetic tape challenging?

What part of the VTR records the signal onto the magnetic tape?

What are the transducers in a VTR?

What two energy forms does a VTR transduce between?

When a magnet moves past an electrical conductor, what happens?

When an electrical current moves through a conductor, what type of energy is radiated?

What advantages did the invention of a VTR bring to the video industry?

Time-Base Correctors

What is the root of the problem which time base correctors attempt to solve?

What are time-base errors?

What is the purpose of a video reference to a VTR or TBC?

How does a TBC re-time a video signal?

Why must all decks and TBC's have a common video reference signal?

What are some other names for video reference signal?

What does black burst mean?

What does the processing amplifier (proc amp) allow?

Editing

How was video tape first edited?

How does film differ from video?

How is film edited?

How is video editing like dubbing?

What is a video equivalent of a film work print?

What is a window dub?

What use are window dubs in the video editing process?

What is the difference between off-line and on-line editing?

Time Code

What are some advantages to using time code?

What information is stored in the time code signal?

What is the internationally accepted time code standard?

What is the Sony Hi-8 time code standard?

Name several ways that time code can be represented:

What is the difference between VITC and LTC?

What is Drop-Frame time code for?

Write the next time code number to occur in Drop-Frame mode:

22:00:59:29... ____________

14:08:59:29...____________

00:09:59:29...____________

23:59:59:29... ____________

When is drop-frame time code useful?

Digital

What is the most fundamental piece of information a digital computer can process?

What is the difference between analog and digital?

What makes a device digital?

What is an A-to-D?

What's the use of an A-to-D converter?

What does quantizing mean?

Which would be more accurate, 4-bit or 8-bit quantization, and why?

What is sampling?

What is the sampling rate of a CD?

What is the quantization rate of CD?

What are the sampling rates for current digital video signals (CCIR 601)?

Speed of digital systems vary widely. What's a typical computer processing speed you know?

What is compression and how does it work?

What's the difference between lossy and lossless compression?

Computers

What can computers do?

How do computers make decisions?.

The decimal system is based on what value?

The binary system is based on what value?

One bit represents how many states?

Two bits represents how many states?

Eight bits represents how many combinations?

24 bits represents how many combinations?

What is hexadecimal?

Translate 10010111 (binary) to its decimal equivalent:

Translate 200 (decimal) to its binary equivalent:

A byte is how many bits?

What is a digital word?

What sort of information can be represented digitally?

LIGHT

A black and white image only makes use of one characteristic of light. Which one is it? brightness

A car that is described as rich red as opposed to pale red is describing which characteristic? saturation

A blue book versus a green book versus a yellow book is defined by which characteristic of light? hue

A white object has how much saturation? none

A black object has how much brightness? none, little

A white object has *which* hue? none, all

A black object has*which* hue? none, all

A black object has how much saturation? none

The first thing light encounters in a video system is the lens.

A lens serves what purpose? to gather and focus light

What would be missing in a video or vision system without a lens? the strength of the light getting inside the camera or eyeball would be much less, so a less strong image would be produced

How does a mirror affect light? It reflects all light.

Modern light theories state that light behaves how? As both particles and as waves.

Examples of light emitting objects are: TV, computer screen, fireflies, stars, LEDs, light bulbs, phosphorescent, glow-in-the-dark things.

Examples of reflected light are: cat's eyes, the moon, any non-light emitting object that you can see.

An object that is photoemissive is an object that emits, or produces, light energy.

Describe the Additive Color Theory: adding colors to each other produces other colors, and the addition of all colors produces white. The lack of all colors is black.

Sending white light through a prism will produce what at the other end? a spectrum, or rainbow, of all colors.

Combining all the colors of the rainbow into a prism will produce what? White light.

What are the three primary colors in modern video systems? Red, Green, Blue.

What are the three characteristics of light that the eye can differentiate? Hue, Saturation, Brightness.

In Additive Color Theory, Yellow is a combination of RED and GREEN.

Magenta is a combination of RED and BLUE.

Cyan is a combination of BLUE and GREEN.

Black is the lack of all light or lack of all three primary colors.

White is the presence of all three primary colors, equal, at highest magnitude.

A filter does what? Absorbs of light frequencies except the one that the filter is made to pass.

A Yellow filter passes what color frequency of light? Yellow.

And it absorbs which color frequencies of light? It absorbs all frequencies except yellow

Electromagnetic Radiation

Define frequency: the number of times something oscillates, usually per second.

What is a Hertz? The number of times something oscillates in one second... frequency.

Define the following prefixes:

kilo-
Mega-
Giga-
Tera-

centi-
milli-
micro-
nano-
pico-

X-Rays, light, microwaves, gamma rays, and radio waves are all part of the electromagnetic spectrum.

What type of radiation is at the "slow" end of the frequency spectrum? radio waves.

What type of radiation is at the high-frequency end of the spectrum? X-rays, gamma rays.

Which color has a higher frequency, red or violet? Violet.

Which color frequency is just below the range of visible light? Infra-red.

What medium does sound travel in? Air.

What medium does light travel in? Electromagnetic fields.

What is the frequency response (range) of human hearing? 20 to 20,000 Hz.

Name the main color distinctions of the rainbow: Red, orange, yellow, green, blue, indigo, violet.

What three characteristics distinguish one color from another? Hue, saturation, brightness.

What determines the color of an object? Wavelengths or frequencies of light, and the human eye.

Why can't we see X-Rays? Because our eyes are not sensitive to electromagnetic radiation above the range of visible light.

Why don't humans hear radio waves *before* they are picked up by a radio? Because our ears are not sensitive to electromagnetic radiation, only pressure variations in air. Radios strip away the electromagnetic carrier wave and then drive a speaker which pushes air according to the strength of the transmitted signal.

Radio Frequencies and Modulation

What is the carrier of information in radio / television broadcast? Electromagnetic radio wave.

What is the difference between light, microwaves, X-rays and radio waves? Frequency.

What do X-rays, radio waves, light waves and microwaves have in common? They are all electromagnetic radiation.

How can radio waves be used to transmit information? A signal containing information can modulate the carrier wave. The carrier frequency can be stripped away at the receiver and all that remains is the original information.

What is modulation? Altering a carrier medium according to some information signal.

Describe amplitude modulation: the carrier frequency is unaltered in amplitude modulation, but the amplitude varies with the strength of the signal.

Describe frequency modulation: the carrier frequency is increased or decreased dependent on the strength of the original signal.

What is the range of carrier frequencies used in amplitude modulation? 540 to 1600 kHz, every 10 kHz.

What is the FM range of carrier frequencies? 88 to 108 Mhz, every 0.2 MHz.

What is the frequency range of light radiation? 375,000 to 800,000 Ghz.

Electricity

List some examples of electricity that occur in nature: Lightning, lightning bugs, static electricity.

All matter is made up of atoms.

Name the subatomic particles: proton, neutron, and electron.
Specify their charge: positive, neutral, negative.

What causes electricity? Voltage, or electrical potential between two points. When their is a difference in charge between two points, their is potential for electrons to flow from point to another.

What does electrical potential mean? The difference in charge between two points, causing a force between them.

What is another name for electrical potential? Voltage.

What is electrical current? The number of electrons passing a certain point in one second.

What is another name for electrical current? Amps, or amperes.

A strong voltage will create a strong or weak current? Strong.

A weak current implies a strong or weak voltage? Weak.

If there is a high current flowing through a wire, what does this mean about the two terminals (or ends) of the wire? There is a potential electrical difference between them, one has more or less electrons than the other.

What blocks electrical current from flowing efficiently? Resistance.

What causes electrical current to be diminished in a material? The molecular structure, lack of free electron flow.

What is an electrical conductor? A material that allows electrons to flow freely.

What is an electrical insulator? A material that prevents electron flow.

What is the unit of measurement for electrical resistance? Ohms.

What is Ohms's Law? Voltage = Current x Resistance.

What does Ohm's Law convey? The relationship between electrical potential (voltage), current and resistance.

Do batteries produce alternating current or direct current? Direct current.

What is the difference between alternating current and direct current? Direct current is at a constant voltage, so the flow of electrons is constant. The potential difference between two points does not alter. Alternating current is caused by a continuously fluctuating electrical potential between two points, so the current alternates.

What are the advantages of alternating current? AC travels long distances and can be used to represent signals like light and audio. AC signals can be modulated to represent information.

What are advantages of direct current? Most circuits in cameras, TV's, computers, are based on DC. DC is necessary in most electrical circuits.

What are the AC specifications (frequency, peak voltages) in the United States? 60 Hz, 120 Volts (average).

What is the British AC standard? 50 Hz, 220 Volts.

What is a name for resistance to an alternating current? impedance.

What are possible means of getting power to a video camera? DC - batteries. AC-to-DC adapter.

Transduction

What is transduction? Proportionally turning one form of energy into another. The entire energy of a system is always conserved.

Name several types of audio transducers: ear- acoustic energy to electrochemical activity. Speaker- electrical energy to acoustic energy.

Describe several forms of visual transducers: eye- light to electrochemical brain activity. Video camera- light to electrical energy.

What is useful about transduction? It allows information to be represented in a form more suited for processing: light as electricity can be processed with electrical technology instead of optical technology.

In the case of audio and video technology, transducing energy into electrical energy is only for processing. What must happen to the signal for final listening or viewing? The electrical energy must be transduced back to light and sound.

Time

Name some ways to detect the passage of time: Clock ticks, the position of the sun, a clean room gets messier, a young child becomes an adult.

Why does a bicycle wheel or a horse's legs tend to blur when moving quickly? Because human eyes are not time-sensitive enough to detect minute changes through time. Eyes take a relatively long time to adjust to photons sensitizing them.

The time it takes for the earth to orbit the sun is called one year.

The time it takes for the moon to orbit the earth is called one month.

A day is a measurement of what cycle? The earth revolving around its axis.

How many seconds are in a year? 365 days x 24 hours x 60 minutes x 60 seconds = 31,536,000 seconds.

How many picoseconds are in one second? One trillion.

Do things happen in less than a second? Yes.

Do things happen in less than a trillionth of a second? Yes.

How many lines are in an NTSC video frame? 525 lines.

How many NTSC video frames are in a second (approximately)? 30 frames.

How many lines per second in an NTSC video signal? 15,750 lines per second.

What is the frequency of lines (per second) of NTSC video? 15,750 Hz.

Could video lines be considered a measurement of time? Yes.

How long does it take to draw one line of NTSC video? 1/15,750 second.

How many NTSC video lines are in a day? 1,360,800,000 NTSC video lines per day.

Video

What is the purpose of using video? To communicate light information between two points. To communicate moving images between two points.

What is video? Light information converted to electricity in real, discrete intervals of time, then converted back to light.

What are some advantages of video (over film)? It can be immediately viewed. It can be processed real time and with all of the latest electrical breakthroughs applied. It can be broadcast from one point to many points immediately. It can be stored for many years on a relatively inexpensive medium (magnetic tape).

A video camera attempts to mimic what natural phenomenon? Human vision.

What does transduction mean? Converting one form of energy to another, maintaining the relative levels.

What is a photoelectric material? A material which transduces electrical energy (electrons) into light energy (photons) and vice-versa.

How does a video tube (transducer) convert light into electricity? By keeping a consistent voltage (or electrical charge) at one end of the tube, there is a relative potential change depending on the number of photons hitting different points on the surface of the tube. The photoelectric coating causes photons to be transduced to electrons in varying amounts.

which alters the voltage from each point on the tube surface to the back of the tube. These varying voltages represent the varying brightness information of an image.

How is an entire image (many points of light) transduced to an electrical signal that can only represent one luminance level at a time? By scanning a beam of electrons at a constant voltage across the screen, varying voltages between the beam and the surface of the tube are produced. These voltages, as they are scanned, are simultaneously output as a varying electrical signal. It is the scanning that makes it possible for the signal to represent only one portion of the image at a time.

How quickly can human eyes detect change? 24 frames a second, maybe a bit more.

What is the purpose of scanning with an electron gun? To serially represent all the brightness values of an image onto one electrical signal.

How is the electron beam aimed? Using two magnetic fields: horizontal and vertical.

Why does the electron gun shut off after scanning a line of video? To prevent a re-trace of electrons and light when the gun must move to the next line. A faint diagonal line would appear beneath every video line otherwise.

What is horizontal and vertical blanking? The time after each line (horizontally) and after each field (vertically) that the electron gun is shut off.

Scanning first the odd lines and then the even lines is called what? Interlace scanning.

How many electron guns are in a color video camera or monitor? Three: R, G, B.

Color Video Signals

What determines the color of an object? The human eye. The wavelength of the light being reflected or absorbed from an object.

Light is part of the electromagnetic spectrum.

What is the additive color theory of light? All wavelengths or colors of light combined produce white and a lack of all light will be black.

What are the three primary colors in video? Red, Green, and Blue.

What is the transducer in a video camera? The video tube.

What does the video camera transducer do? Transduces light energy to electrical energy over time.

What does a video monitor transducer do? Transduces electrical energy (video signal) to light energy.

What device produces the scanning behavior of video signals? The electron gun.

What is the purpose of scanning? It is a way of sending all of the information in a picture frame on an electrical signal that can only represent one brightness value at a time.

Why aren't Red, Green and Blue signals broadcast? Its' too much information, we would need three channels.

In what ways are black-and-white and color TV signals compatible? A black and white TV can play a modern color signal (in black-and-white) but a color TV would be lacking information from a black-and-white signal.

What wavelength (or color) of light do humans perceive the highest detail? Green.

Colorimetery is the science of color and human perception of color.

What is the RGB to luminance (Y) equation? Y = 59% Green, 30% Red, 11% Blue.

Which of the three signals R, G, and B does a black-and-white television use? None of them, unless you only want to display one color component of an image.

What are the two color difference channels called? R-Y and B-Y.

Name the three color component signals derived from R, G, B: Y, R-Y, B-Y.

With the signal Y (luminance) and R-Y, how can R be derived? R-Y + Y = R.

From Y, R and B, how is G derived? Y - (30%R - 11%B) = 59%G. Increase Green to 100%.

How is Y/C derived from Y, R-Y and B-Y? R-Y and B-Y are combined to form C.

How is NTSC composite derived from Y/C? C is modulated on a 3.58 Mhz subcarrier and mixed with the Y (luminance) information.

Processing RGB signals to NTSC composite is known as encoding the video signal.

Deriving RGB from the NTSC composite signal is known as decoding the video signal.

A color video monitor has how many electron guns? Three: R, G, B.

Video - the Monitor

What is the name of the transducer in a video monitor? Cathode ray tube (CRT).

What does the video monitor transducer do? Transduces electrical video signal into light.

What allows a camera and a monitor to scan their electron guns in time with one another? Horizontal and vertical synchronization pulses.

What are some problems if the camera and monitor electron guns are not scanning together, in sync? The picture will not be accurately reproduced, a vertical or horizontal roll will occur. The lines will be randomly scanned across the screen and a coherent image will not be reproduced.

Dubbing

What type of video input/output does a BetacamSP deck have? Component (Y, R-Y, B-Y), composite.

What video input/output does a Hi-8 deck have? S-video (Y/C) and composite.

Name two types of video input/output for 3/4" U-matic decks: Dub (Y/C-688) and composite.

What is the CTDM signal used for? Dubbing BetacamSP to BetacamSP.

What devices use component serial digital video connections? Quantel, D-1, Digital Betacam, digital video switchers, DVE's.

What kind of connector is used for professional audio? XLR.

What kind of audio connectors are used on consumer VHS decks? RCA.

What is the AES/EBU audio standard for? Professional digital audio.

What is the name of the consumer digital audio standard? S/PDIF.

Name several types of remote connectors: 9-, 33-, and 45-pin.

Name two remote device communications protocols used between decks and edit controllers: RS-232 and the more modern RS-422.

What are the typical tracks on a video tape? Control track, two longitudinal audio tracks, video track, time code (address track), sometimes Hi-Fi AFM audio.

What is the control track for on a tape? To ensure the VTR will play back the video tape at the same speed it was recorded. It's like a series of electronic sprocket holes laid down when a deck is in record mode, and gives an indication of the speed the recording deck is playing.

From Hi-8 to BetacamSP, what transcoding of signals would yield the highest quality? S-video (Y/C) to CTDM or Y, R-Y, B-Y.

Dubbing 3/4" to VHS would require what kind of video connection? Composite.

Most modern decks use which number of pins on remote cables? 9-pin.

What is the current protocol for deck communications? RS-422.

To connect consumer VTR audio to professional audio means what three things are necessary? Impedance matching, unbalanced to balanced signals, and connector adapting.

NTSC Video

NTSC stands for National Television Standards Committee.

What is the purpose of the NTSC television standard? It is a standard enabling different manufacturers to produce video and broadcast equipment which will be compatible trhoughout the U.S.

Define the following NTSC specifications:

Fields per frame: 2
Frames per second: 30 (black and white), 29.97 (color)
Fields per second: 60 (black and white), 59.94
Lines per field: 262.5
Lines per frame: 525
Voltage of signal (peak to peak): 1 volt
IRE units of signal (peak to peak): 140 IRE
AMPLITUDE of sync pulse: 40 IRE
AMPLITUDE of color burst: 40 IRE
Blanking level: 0 IRE
Black level: 7.5 IRE
Peak white level: 100 IRE
Color subcarrier frequency: 3.58 Mhz.

Horizontal synchronization pulses occur how often? Once per video line.

Vertical synchronization pulses occur how often? Once per video field.

Where in the NTSC video signal is the color burst located? In the horizontal blanking, after the horizontal sync pulse.

Is NTSC a composite signal or a component signal? Composite.

What is the difference between a composite and a component signal? Composite signal is a sum of the luminance and color channel. Component keeps the luminance and the chrominance separated.

What are some disadvantages to NTSC? Composite signal means luminance and chrominance interfere with each other. NTSC is only 525 lines instead of PAL 625 lines.

What's the advantage of a composite signal? One wire, one transmission channel, less bandwidth.

How is NTSC video encoded from the original R, G, B color signals from the video camera? RIB is sent through a matrix to derive Y, R-Y, B-Y; R-Y and B-Y are processed in a matrix to give C. Y and C are combined to produce composite NTSC.

Video Tape Recorders

When was the video tape recorder invented? 1955.

What makes recording a video signal on magnetic tape challenging? There is a lot of information to be stored on a small amount of space.

What part of the VTR records the signal onto the magnetic tape? The record head.

What are the transducers in a VTR? The record and playback heads.

What two energy forms does a VTR transduce between? Magnetism and electricity.

When a magnet moves past an electrical conductor, what happens? An electrical current is created.

When an electrical current moves through a conductor, what type of energy is radiated? Magnetic energy.

What advantages did the invention of a VTR bring to the video industry? Shows no longer had to be live. Images could be stored and replayed later. Electronic video editing is possible.

Time-Base Correctors

What is the root of the problem which time base correctors attempt to solve? Video decks cannot accurately playback all of the video signal in the right time relationship to how the original video signal was constructed.

What are time-base errors? Errors that happen during VTR playback in which a motor does not play precisely the same speed at which the signal was recorded or signals coming off of the tape are not in accurate phase relationship with one another.

What is the purpose of a video reference to a VTR or TBC? To give a deck a means of reconstructing an accurate signal from the imprecise one playing off of the tape.

How does a TBC re-time a video signal? It stores up to a frame of video at a time and either delays or advances the signal output depending on the input signal's timing relationship to the reference signal.

Why must all decks and TBC's have a common video reference signal? It is a way of ensuring that all video signals leaving the decks will be in time with one another. This will allow the to be mixed together using a video switcher, for example, without an horiztonal or hue shifting.

What are some other names for video reference signal? Master clock, black burst, synchronization, house sync.

What does black burst mean? 7.5 IRE video signal with consistent -40 IRE sync pulses.

What does the processing amplifier (proc amp) allow? Adjustment of video black, white, chrominance and hue. Using the proc amp helps to maintain broadcast quality video output.

Editing

How was video tape first edited? With a splice, just like film. A magnetic ink was placed on the tape and a microscope was used to find the tracks. Then it was cut with a razor and spliced.

How does film differ from video? Film is a chemically processed emulsion medium. Video is magnetic based median. The silver particles on the film transduce amounts of light to chemical activity. Magnetic tape stores the magnetic fluctuations caused by an electrical video signal.

How is film edited? By hand, cutting between frames and then splicing.

How is video editing like dubbing? Because you are simply copying video signals from one tape (the source tape) to another (the master tape). Only selected portions of the video footage are transferred.

What is a video equivalent of a film work print? A window dub or a safety dub.

What is a window dub? A dub of video original video footage reflecting the time code information on the tape. The time code is visually burned into a "window" (or tiny rectangle) on the video picture.

What use are window dubs in the video editing process? They act as a means of representing the original footage and referencing it when necessary. They are also cheaper to work with because they are on more inexpensive formats.

What is the difference between off-line and on-line editing? Off-line editing is the creative, cheap format cut. The on-line edit is the final master edit using the time code numbers attained during the long off-line process. Video signals placed on the final tape are closely watched to maintain broadcast quality.

Time Code

What are some advantages to using time code? Every frame of video can be uniquely identified and located on a video tape.

What information is stored in the time code signal? Hours, minutes, seconds, and frames. Also user-bits can be stored within the signal.

What is the internationally accepted time code standard? SMPTE.

What is the Sony Hi-8 time code standard? RC time code.

Name several ways that time code can be represented: LTC, longitudinal time code, as audio information. VITC, vertical interval time code as video information.

What is the difference between VITC and LTC? VITC is bright and dark spots (high low voltages) in the vertical interval of the video signal and can be seen when a deck is in pause. LTC is an audio signal undetectable in pause and can be read backwards and forwards. LTC is written on an audio track of a video tape.

What is Drop-Frame time code for? Because the frame rate of NTSC video was adjusted slightly to 29/97 frames per second, the time code counters became inaccurate. TO cover for this error, 2 frames are dropped each minute except the tenth minute. This keeps long term time errors to occur.

Write the next time code number to occur in Drop-Frame mode:

22:00:59:29...22:01:00:02
14:08:59:29...14:09:00:02
00:09:59:29...00:10:00:00
23:59:59:29...00:00:00:00

When is drop-frame time code useful? When doing a commercial or PSA where each frame counts.

Digital

What is the most fundamental piece of information a digital computer can process? One bit, having two states: on or off, 0 or 1.

What is the difference between analog and digital? Digital codes, processes and stores discrete amounts of information. Analog signals are a continuous.

What makes a device digital? The fact that it processes discrete, on or off, bits of information instead of working on a continuous analog signal.

What is an A-to-D? An analog to digital converter.

What's the use of an A-to-D converter? It translates continuous analog signals of varying voltages into discrete steps all represented as 0's and 1's. It's necessary for a computer to work on any information that it first be digitally coded.

What does quantizing mean? Breaking down infintite possibilities of signal amplitude to a finite number of discrete steps that a signal can be said to be at.

Which would be more accurate, 4-bit or 8-bit qunatization, and why? An 8-bit system will be more accurate to the original signal because there are more bits of information each representing tinier distinctions between two points of an audio signal waveform.

What is sampling? Taking quantization readings at a specific frequency or rate.

What is the sampling rate of a CD? 44.1 kHz.

What is the quantization rate of CD? 16-biit.

What are the sampling rates for current digital video signals (CCIR 601)? 4:2:2:, 13.5 Mhz, 6.25 Mhz for each color .

Speed of digital systems vary widely. What's a typical computer processing speed you know? 120 Mhz, 33 Mhz.

What is compression and how does it work? A means of reducing information in a digital video signal by stripping it down and removing or encoding redundant information.

What's the difference between lossy and lossless compression? Lossless compression means even though information was reduced, it can be attained at the receiving end by means of reconstructin the signal via the reverse mathematical process. Lossy compression removes enough data so that it is not all recoverable.

Computers

What can computers do? Computers are just giant calculators waiting to be assigned a task: digital information can be made to represent information.

How do computers make decsions? Logically, based on true or false.

The decimal system is based on what value? ten.

The binary system is based on what value? two.

One bit represents how many states? two.

Two bits represents how many states? four.

Eight bits represents how many combinations? 256.

24 bits represents how many combinations? 16,777,216.

What is hexadecimal? Base-16 numerical system.

Translate 10010111 (binary) to its decimal equivalent: 151.

Translate 200 (decimal) to its binary equivalent: 11001000.

A byte is how many bits? Eight.

What is a digital word? A digital word is a specified number of digital placeholders that represents "words" like synchronization, transmission start and stop codes, etc.

What sort of information can be represented digitally? Just about any kind: audio signals, video signals, text, temperature, mathematical equations, still photographs, music notation, 3-D space design.

"What is video? - electronic representation of moving images. **Light** represented as electricity over time."

Light - "something that makes vision possible"

Light is the fundamental energy form used in video. It is a form of energy defined by our eyes. If you look at something, you are allowing light into your eyes. An eye does the job of taking light energy and transforming it into electrochemical energy which your brain interprets. You do not need to know anything about what light *is* to see. Seeing just happens.

Let's Build an Eyeball

Oh, to build a perfect eyeball!... of course, a perfect eyeball would be one that sees just as well as our own eyes. A perfect eyeball would take light energy and turn it into electrochemical brain activity just as our own eyes do.

First, we need to define characteristics of light that the eyeball distinguishes:

Brightness: when something emits a lot of light, we call that bright. A lack of light, or low levels of light we call dark. Look at this scene. It is made up of a range of brightness values...

Hue: *which* color an object is. The human eyeball can differentiate between something that is *red* and something that is *blue* , for example.
{show red something vs. blue something vs. green something}

Saturation: *How much* of a particular Hue is known as the Saturation of that Hue.
{show a lot of red vs. not a lot of red vs. no red... high brightness, low brightness}

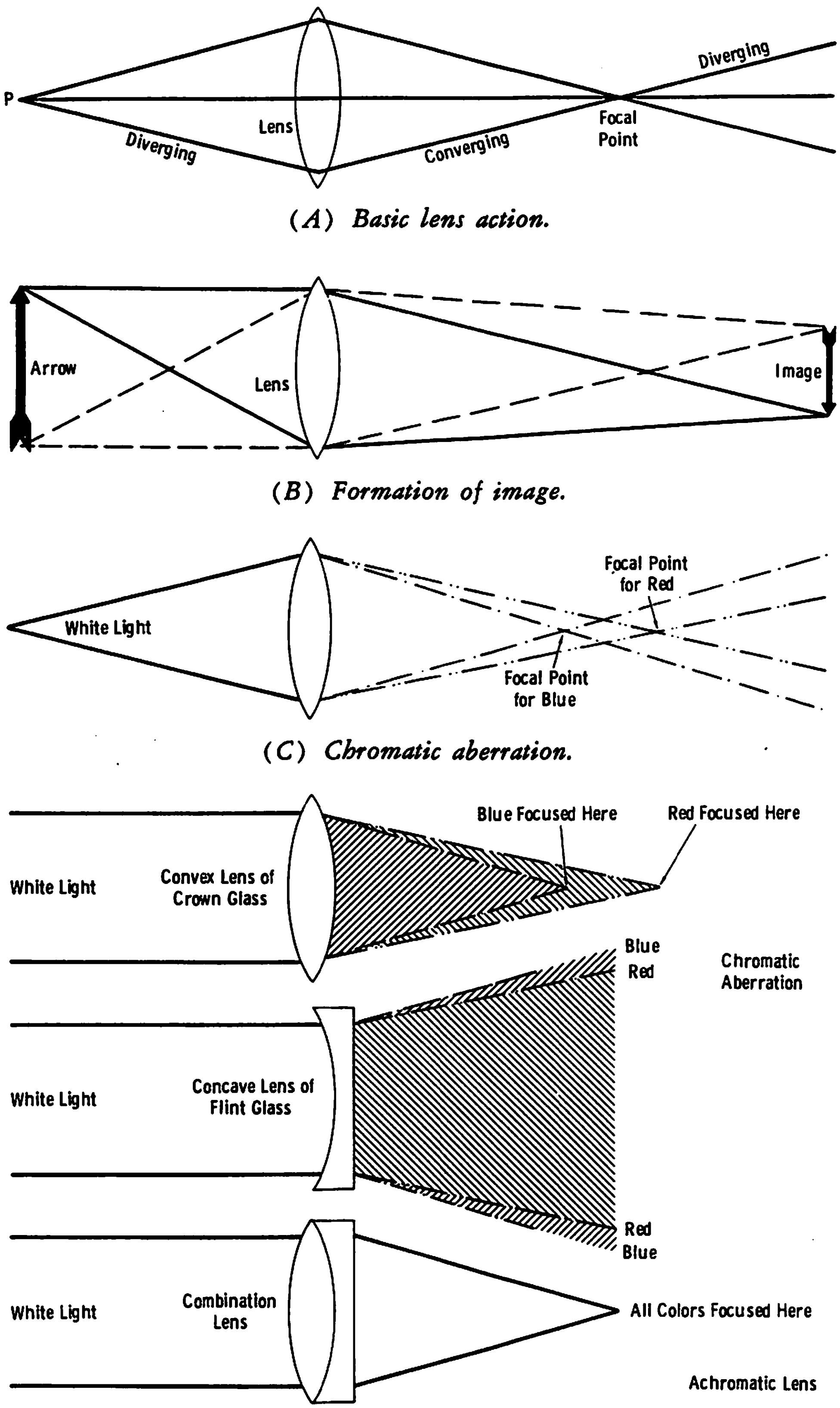
So, the three characteristics of light we are concerned about are Hue, Saturation and Brightness (H, S, and B).

Brightness is enough to describe a black-and-white image, and so it is sufficient to describe a basic video system. Early video, like photography, was black and white because there is simply less information to worry about. In a black-and-white video system, the only thing to keep track of is *how bright* an image is at any given time.

Let's Build a Video Camera

The human eyeball is spherical. The first thing light encounters is the lens at the edge of the eye. The purpose of the lens is to gather and focus light into the eyeball. It is the light collector, taking it all in indiscriminately.

The first thing we'll have in our video camera is a lens. The lens gathers and focuses light into the video camera. The purpose of any lens is to collect and focus light. Because it is curved, it brings all light entering it to a focused point.



(*A*) *Basic lens action.*

(*B*) *Formation of image.*

(*C*) *Chromatic aberration.*

(*D*) *Achromatic lens.*

Fig. 1-3. Development of achromatic lens.

What is Light?

There are many theories about how light actually behaves. But what is light, really? It doesn't have any form or structure that we can plainly see. In fact, what we see is light, not the structure of how it works. It's like looking at a congested highway and not noticing that beneath the hulls of the cars is really spinning engine pistons, electrical systems, cooling radiators, etc. Light is what we see, not the actual objects that the light sources from.

For example, if you are reading this page under some light source, like a lamp, what you are really seeing is the light bouncing from the lamp to the paper to your eye. The black letters on this paper, unlike the white page, absorb all light that hit them, so no light is reflected into your eye, so you *don't* detect any light where there are letters.

The detection of all reflected light is usually called white.

The detection of no reflected light is called black.

If you hold this page under a yellow, tungsten lamp, you will see a yellow page. This is because the white page is a perfect reflector. It filters out no light. Practically all the light leaving the tungsten lamp bounces from the page to your eye.

The black ink that forms the letters absorbs all light, no matter what color it is. As a result, there is no light entering your eye. This is called black, or a lack of light.

What you see is the *effect* that different materials have on light. Some materials reflect all light, some filter certain colors, some absorb all light. Some objects reflect light perfectly, like mirrors.

How Light Behaves

One theory states that light behaves just like waves in the ocean. This is the wave theory of light. There is another theory that says that light behaves like physical particles, like subatomic particles. This is the particle theory of light. Modern theory states that light behaves BOTH like waves and like particles. The theoretical name for a particle of light is called a *photon.*

Agggh! Radiation

Light is radiation. Light is radiated, or emitted, from certain objects. The sun, for example, is a gigantic light emitter. So is a light bulb, or more specifically, the filament inside the bulb. Certain materials and chemicals are light emitters. Others are light absorbers. An object that emits light is called *photoemissive*. A TV screen, light bulb, star, or a firefly, for example, is photoemissive. These objects, by nature, emit light. A piece of paper or a red book that you may see in a room is due to reflected light. All reflected light initially comes from emitted light.

Additive Color Theory of Light

As mentioned before, there are a lot of ways to describe how light behaves. One incredibly practical model is called the Additive Color Theory of Light.
The most important aspect of this theory is that WHITE light is the presence of all colors (all frequencies) at one moment. By sending white light through a prism and refracting the light (bending it at different angles dependent on the frequency), an entire rainbow of light is produced. This rainbow is the component colors of white light, spread out by the prism. WHITE light, then, is the presence of all colors. No light at all is considered black. And any specific color is simply a specific frequency of light energy.

The Additive Color Theory describes color in a way that implies adding one color to another produces yet a third color, and so on. It states that three primary (fundamental) colors can be chosen upon which all other colors can be created from mixtures of the primaries. In video and computer systems, the three primary colors that were chosen were: RED, GREEN, and BLUE.

A purely red object is simply emitting or reflecting RED light and absorbing GREEN and BLUE.
A purely GREEN object is simply emitting or reflecting GREEN light and absorbing BLUE and RED.
Likewise with a BLUE object.
BLACK is the lack of all colors. BLACK is the result of no light being emitted, or all light being absorbed into the object.
WHITE, conversely, is the presence of GREEN, BLUE, and RED at equal magnitude.
MAGENTA, a near cousin of violet, is the combination of BLUE and RED.
CYAN, a sort of aqua blue, is the combination of BLUE and GREEN.
YELLOW is the combination of RED and GREEN.

Filters

Filters are nothing more than objects that absorb specific frequencies of light energy and allow others to pass. When you look through a RED filter, for example, you tend to see everything as red. This is because all color frequencies of light are absorbed by the filter EXCEPT the RED frequency. This passes through the RED filter and so RED is what you see. To call it a RED FILTER is not quite accurate because what it is actually doing is not letting any light through other than red. The same is true of any colored filter.

Electromagnetic Radiation

Light is nothing more than energy being radiated, or emitted at a certain *frequency*. Frequency describes how often something happens in one moment in time, usually one second. If I run fast, my legs may hit the ground at a frequency of four shoes per second. If I run slower, the frequency may be two shoes per second.
How often the peak of a wave passes a certain point, or is emitted from an object, is called the frequency of that wave. A wave that fluctuates four peaks per second is faster than one that fluctuates two peaks per second. A faster emission of peaks means the wave has a higher frequency.
How many times something happens in one second is measured in a unit called Hertz (Hz). If I clap my hands once every second, the frequency of my hand clap is one Hertz. If I clap ten times per second, the frequency is ten Hertz.

Light *waves* vibrate at frequencies between 300 and 800 trillion times per second, or 300 to 800 TeraHertz.

1 Hertz	Hz	1 cycle per second	
1 kiloHertz	kHz	1000 times per second	thousand
1 MegaHertz	MHz	1,000,000 times per second	million
1 GigaHertz	GHz	1,000,000,000 times per second	billion
1 TeraHertz	THz	1,000,000,000,000 times per second	trillion

1 centimeter	cm	100th of a meter	hundredth
1 millimeter	mm	1000th of a meter	thousandth
1 micron	mm	1,000,000th of a meter	millionth
1 nanometer	nm	1,000,000,000th of a meter	billionth
1 angstrom unit	A	10,000,000,000th of a meter	ten billionth
1 picometer	pm	1,000,000,000,000th of a meter	trillionth

This means that light vibrates very, very rapidly. Types of energy can be distinguished by the frequency domain in which they vibrate... for example, light is in the domain of 300-800 TeraHertz. Sound is in the range of 20 Hz - 20,000 Hertz (or 20 kHz). Radio waves are in the range of...

This entire spectrum of energy vibrating at different frequencies is called the **electromagnetic spectrum**.
The electromagnetic spectrum is the entire range of wavelengths or frequencies of electromagnetic radiation from the shortest gamma rays to the longest radio waves, visible light comprising only a small part of the range.

Frequency Ranges of Electromagnetic Spectrum

Radio and Microwave Frequencies

Very Low Frequency	VLF	10 - 30 kHz
Low Frequency	LF	30 - 300 kHz
Medium Frequency	MF	300 - 3000 kHz
Amplitude Modulation Radio	AM	540 - 1600 kHz, every 10 kHz
High Frequency	HF	3 - 30 MHz
Citizen's Band	CB	26.965 - 27.405 MHz
Very High Frequency	VHF	30 - 300 MHz
Frequency Modulation Radio	FM	88 - 108 MHz, every 0.2 MHz
Ultra High Frequency	UHF	300 - 3000 MHz
Super High Frequency	SHF	3000 - 30,000 MHz
Extremely High Frequency (microwaves)	EHF	30,000 - 300,000 MHz

Visible and Invisible Light Frequencies

Infra-red light	IR	300,000 MHz - 375,000 GHz
Visible light		(around 600,000 GHz)
Red		375,000 GHz
Orange		
Yellow		
Green		
Blue		
Indigo		
Violet		800,000 GHz
Ultra-violet light	UV	100,000,000 GHz

X-Rays		3,000,000,000 GHz

Gamma and Cosmic Rays		> 300,000,000 GHz

It should be noted that sound does not appear anywhere in the electromagnetic spectrum because it is not electromagnetic by nature. Sound is the result of air varying in pressure at different frequencies. Light is the result of electromagnetic radiation varying at different frequencies.

Humans detect certain ranges of varying air pressure frequencies and electromagnetic frequencies. This is not true for all creatures and materials. Bats, for example, detect energy in a higher-than-sound frequency range called ultrasound. And certain chemicals and film emulsions are sensitive to infrared light frequencies, meaning below the frequency range of light (specifically red). For almost every frequency range there is some emitter and some detector for that range.

Frequency Range	Emitter or Detector	Energy Medium
20 - 20,000 Hz	Human Ears	Air
20 - >20,000 Hz	Dog, cat hearing	Air
>20,000 Hz (ultrasound)	Bats	Air
1000 kHz	AM Radio	Electromagnetic
100 MHz	FM Radio	Electromagnetic
30,000 MHz	Microwave ovens	Electromagnetic
300 - 800 TeraHz	Human, animal eyes	Electromagnetic
3,000,000 TeraHz	X-Ray machines, film	Electromagnetic

Wavelengths and Frequencies Determine Color

So what really distinguishes between red and blue? Light doesn't. Your eye does. Light entering your eye which vibrates at approximately 300 trillion times per second is detected by certain detectors in your eye. The brain registers the detection and determines that the color is RED. If a light entering your eye which vibrates at approximately 800 trillion times per second is detected, the brain will interpret that to mean VIOLET or PURPLE. In between is the range of *visible light*: red, orange, yellow, green, blue, indigo, violet.

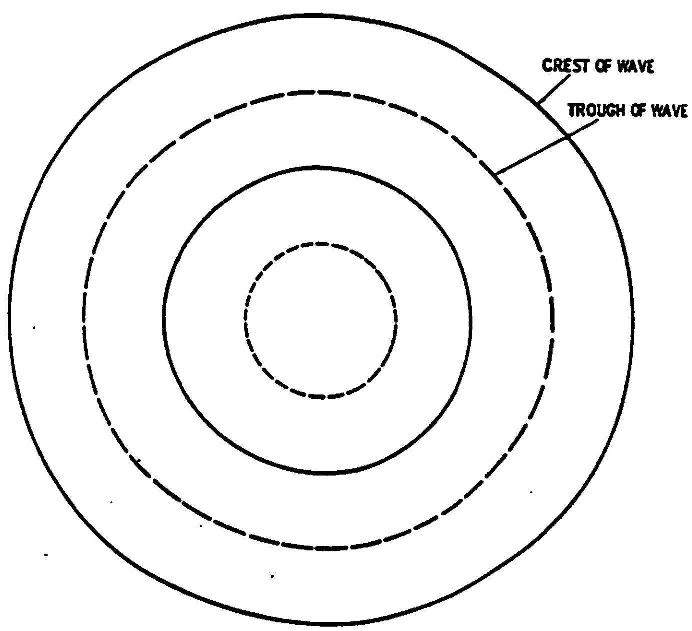
ROY G BIV

Color is determined by how rapidly the wave is vibrating, just like sound. Just like in musical sounds, differing frequencies mean different pitches, differing light frequencies determine different colors.

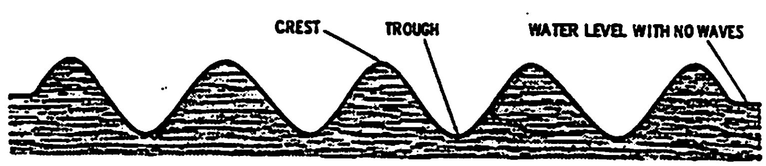
Looking at a red light means looking at light with a lower frequency of vibration than looking at a blue or violet light.

WHAT IS SOUND?

Sound is created by the movement of molecules moving through the air in waves similiar to the waves in a pool of water into which a stone has been thrown. Sound waves are generated by a moving body in contact with the air. This could be someone's vocal chords or a string of a quitar. The motion of the generating body produces changes in the air pressure in proportion to both the FREQUENCY and the AMPLITUDE of its vibrations. Sound travels at a constant speed of 1130 feet per second, or 340 meters per second.



(A) Top view.



(B) Side view.

Fig. 1-1. Waves created by a stone falling into a pool of water.

CHAPTER 1: WHAT IS SOUND?

Sound is a word we use to refer to vibrations that we're able to hear. By examining the nature of sound vibrations, we'll be able to understand the concepts of:

- frequency
- sound waves and pressure
- the audio spectrum
- pitch

and several other important terms which are used in multi-track recording. These concepts may seem a bit theoretical, but much of what follows is essential for understanding everything from proper microphone placement to adjusting levels.

Vibrations are everywhere, from the movement of a bee's wings to a truck rolling by on the street. Some vibrations which can't be heard can still be felt—as anyone who's experienced an earthquake will appreciate. And then there are vibrations which our senses cannot detect, such as radio frequencies.

We're able to describe audible vibrations in terms of how loud they sound, whether they are 'high' or 'low' in pitch, and in terms of the character (timbre) of sound, such as 'harsh,' 'bell-like,' 'clean,' and other subjective adjectives. Let's take a look at the components of sound which distinguish one from another.

Frequency

Vibrations travel through space—air or solid—in the form of *waves*. Imagine what happens when a pebble is dropped into a pond. From where the pebble enters the water (the *source*), ripples, or small waves, form and spread in all directions. Each ripple travels in *cycles*: One complete wave motion, start to peak to trough to start position again, is one cycle.

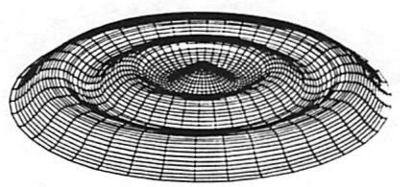


Figure 1.1 Like ripples in a pond, sound travels in waves.

The speed at which a wave completes a cycle is known as its *frequency*. This frequency is expressed as *cycles per second (cps)*, or more commonly, *Hertz (Hz)*, named after the German physicist who first described audible vibrations as sound waves.

Sound Waves And Pressure

When something vibrates in air it creates a series of wave cycles. With each cycle, the air particles will move back and forth, alternately raising and lowering air pressure.

Think of the bee's wings. The wings move back and forth, or *oscillate*, creating waves in the air. If it's your average-size bee, the wings may oscillate at 250 times per

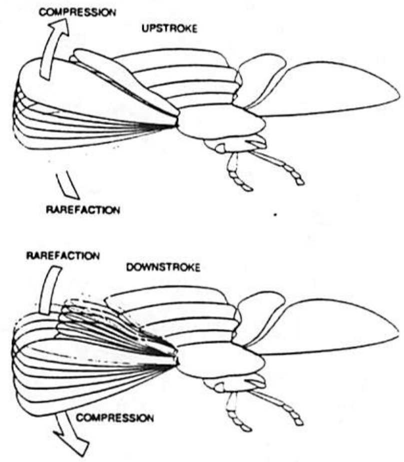


Figure 1.2 Sound waves are created by compression and rarefaction of air.

second. In other words, the *frequency* of the bee's wings is 250 Hz.

As the wing moves out from the body, it *compresses* the air being pushed away, increasing the air pressure of those particles. The air behind the wings decompresses, or *rarefies*. When the wings return, they compress and rarefy air particles in the opposite direction as before. As we can see in the diagrams, these pressure changes create sound waves. In the case of the bee, the waves travel through the air at 250 Hz and reach our ears. At that point, the ear drum, and eventually in turn cilia (tiny hairs in the inner ear), are stimulated 250 times per second, and our brain interprets this stimulation as *sound*.

The Audio Spectrum

The frequency range of human hearing is approximately 20 Hz to 20,000 Hz (20,000 Hz can be expressed as 20 *kiloHertz*, or 20 *kHz*). That is, our ears can detect pressure changes in the air between the rates of 20 Hz to 20 kHz. If something is vibrating at a rate of 20 Hz to 20 kHz, it's creating sound waves, and it can be heard. We call this range the *audio spectrum*.

This range varies, depending upon one's gender and age. Generally females can hear higher—closer towards 20 kHz—as do younger people. Typical hearing for a middle-aged male might be something like 20 Hz to 14 kHz.

Pitch And Frequency Range

When the shape of a sound wave follows a regularly-recurring pattern rather than a random shape, we can say it has a pitch. That is, when a sound source has a specific or dominant frequency at which it's creating sound waves, we can hear a specific pitch. When a sound source is capable of creating (or responding to) more than one pitch, we can say it has a frequency range. Here are some examples of various pitches and ranges:

- The lowest note on a piano is 27.5 Hz, and the highest is 4186 Hz; thus, the frequency range of a piano's note is 27.5 to 4186 Hz.
- The frequency range of the notes of a guitar is 82.41 Hz (low *E*) to 1046.5 Hz (high *C*); a bass guitar's notes range from 41.2 to 523.25 Hz.
- Concert Pitch—the pitch to which orchestras tune—is 440 Hz, the *A* above Middle C on the piano. That's why it's also known as *A-440*. Many orchestras take exceptions to this: For example, the Dresden Symphony tunes to *A-443*, and the Boston Symphony Orchestra has tuned as high as *A-445*!
- The human voice has a range of approximately 80 to 1200 Hz.
- The low buzzing sound that you can hear in an old guitar amp is the frequency of the AC power: 60 Hz (50 Hz outside of North America). The high-pitched tone which emanates from a TV and drives some people crazy: 15,734 Hz.

Frequencies which exist below and above our range of hearing are known respectively as infrasonic and ultrasonic. FM-band radio frequencies emanate from antennae at a range of 88.5 to 108.5 *megaHertz* (million Hertz!); an earthquake can produce frequencies of 5 Hz and lower. And yes, animals have different hearing ranges from humans: Fido's silent dog whistle is usually 25 kHz, and blue whales can communicate at 5 Hz.

In music, a change of an *octave* is an exact doubling or halving of frequency. For example, *A* one octave above *A-440* is 880 Hz, and the *A* one octave lower is 220 Hz.

Wavelength

Anyone who's listened to short-wave radio has heard the term *wavelength*. This is simply another way of measuring frequency, though instead of measuring cycles per second, the actual physical length of the cycle is measured.

Here's an interesting formula:

$$\lambda = V \div F$$

[λ= Wavelength in feet (or meters); V= Velocity of sound in feet (or meters) per second; F= Frequency in Hertz (cycles per second).]

The velocity of sound depends upon the ambient air temperature, but we can average it to be 1,100 feet/second, or 335 meters/second. Using this simple formula, we can figure out that Fido's 25 kHz whistle sound has an individual cycle wavelength of about 1/2" (1.3 cm). Some other sample wavelengths:

- 440 Hz concert pitch = 2.5' (0.76 m).
- Low *E*, bass guitar = 26.7' (8.13 m).
- 5 Hz blue whale call = 220' (67 m)!

* * * *

So far we've learned that:

- Vibrations in the air create sound waves

by changing air pressure.

- Sound waves have cycles, and the frequency of these cycles determines the pitch that we hear when the waves reach our ears.
- A sound's frequency is measured in cycles per second, or Hertz; it can also be physically measured as a wavelength.
- The frequency range of human hearing is known as the audio spectrum.

There are four other important concepts to learn about components of sound waves:

- The *amplitude* of a sound wave determines how loud it sounds to us.
- The *decibel* is the measure of amplitude.
- Sound waves have different *phase* cycles.
- The *timbre* of a particular sound wave—that is, the shape of its cycles—is what makes it sound different from another sound wave.

Amplitude

When we draw a sound wave, we're actually creating a diagram of pressure changes. A peak of the wave represents a compression of air particles, and a trough represents rarefaction. Simply, the greater the distance between peaks and troughs, the greater the *amplitude* of the wave.

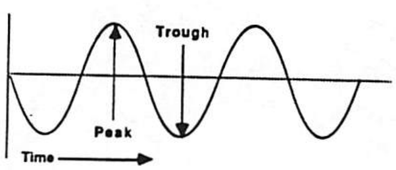


Figure 1.3 The degree of air pressure changes determines the amplitude of a sound.

Amplitude describes the loudness, or volume, of sound. Just as bigger waves crash on the beach with more 'force' than smaller waves, bigger sound waves stimulate our eardrums and cilia with greater force than smaller waves, and sound 'louder' to us. A sound wave which has a greater volume than another sound wave is said to have a greater *sound pressure level* (SPL). This SPL is expressed in terms of decibels.

The Decibel

In this book, we are going to learn about several different types of *decibel* scales. They describe relative levels: sometimes the power levels of audio/electrical circuits, sometimes meter readings of those circuits, and so on.

Decibels are commonly used to express sound pressure levels. With this scale a decibel can be referred to as a *dBA*, or more commonly (though less accurately) as just a *dB*. The 'B' is capitalized, by the way, as a hommage to Alexander Graham Bell, the father of the telephone and other sound-related devices.

Sound pressure is measured on a scale from 0 dB (the threshold of human hearing) about 140 dB (blown eardrums and neral mayhem). But this isn't your average type of scale. For example, the ambient noise in a supermarket may measure about 60 decibels, and a loud rock concert may measure 120 dB. Does this mean your typical rock concert is twice as loud as a supermarket? In fact, no—120 dB is actually *1,000* times louder (in terms of sound pressure) than 60 dB! In fact, a change of just 10 dB represents a doubling of apparent volume, at least to the 'average' ear.

Here's a few more points to keep in mind when discussing sound pressure levels and decibels. First of all, to the 'average' ear, changes of a little as 1 dB are detectible—though some people with otherwise fine hearing require a change as great as 3 dB in order to preceive a difference.

The other point refers back to the fact that for most people, if one sound is 10 dB greater than another, it's twice as loud. That's correct, but only if the two sounds are the same frequency. That's because our ears are less sensitive—especially at relatively quiet levels—to frequencies below 500 Hz and above 7 kHz or so.

An example: Let's say we play a really high note on a guitar, about 1000 Hz, at a quiet level of about 60 dB. In order for the lowest *E* string at 80 Hz to sound just as loud to our ears, we need to play it at about 70 dB. This difference becomes less the louder we play. Of course, none of us run around measuring how loud in dB we play individual notes, but we do compensate naturally as we play quietly. In fact, the 'loudness' control found on many stereos takes these sensitivities into account, by boosting the low and high frequencies at low listening levels. You may, incidentally, run across something known as the *Fletcher-Munson curves*: This is a chart used by audio design engineers, which shows relative loudness of different frequencies at different listening levels. For the most part, fortunately, we don't need to worry about these frequency differences: When sound levels are expresed, it's usually an average of all frequencies.

Phase

Just as ocean waves rise to crests and fall to troughs, so do sound waves. When two identical sound waves rise and fall at the same time and place, they are *in phase*; when they rise and fall at different times, they are *out of phase*.

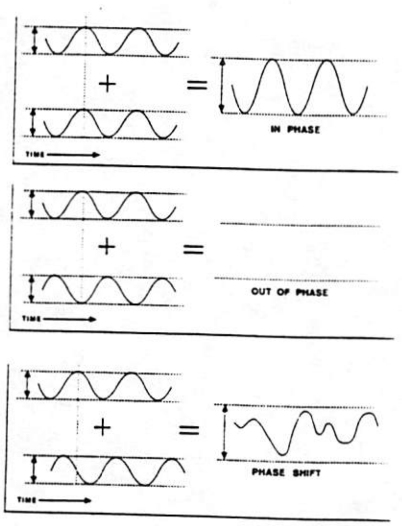


Figure 1.4 Phase addition and cancellation.

In sound (as with water), when waves arrive at a person's ears or a microphone, they add together. Because of this, two identical sound waves which are perfectly in phase will add together to create a *composite* wave which is twice as loud. Two waves which are slightly out of phase have the effect of partially cancelling each other out, however. Audibly, when two waveforms are out of phase with each other, the cancellations create a composite wave with 'holes' in the frequency range. These holes can make the sound seem 'thin,' and can make the source of the sound seem as if it's shifting from side to side. (We'll examine this shifting effect further ahead on page 52).

Finally, if two identical waves which are perfectly 180 degrees out of phase arrive at the same time, they will cancel each other completely, and no sound will be heard!

Timbre And Waveforms

A flute and a saxophone are both able to play a Middle C note—what makes their sound character, their *timbre*, so different from each other?

The answer to that question could fill a book (and has). For our purposes, a simplified overview will help us understand timbre, and conclude our introduction to sound.

Most notes, from an instrument or voice, have one frequency which is the loudest—that's known as the *fundamental* frequency, and it's that frequency which we hear as a note's pitch.

With musical instruments, as with almost all natural sounds, the fundamental frequency is accompanied by ***harmonics***, or *overtones*. These are other frequencies which can be heard—usually at a quieter level—along with the fundamental. These additional frequencies all add together with the fundamental to create a more complex waveform than would be produced by just the fundamental. Their number and relative levels shape the sound's timbre.

Here's another way to look at this: As we've learned, the frequency of a wave determines its pitch. Now we're going to see how the shape of a wave determines its timbre. Let's consider a mathematically 'perfect' wave:

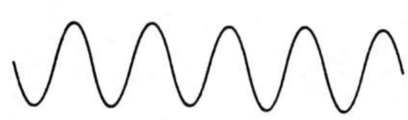


Figure 1.5 A mathematically 'perfect' sine wave.

This wave has smooth, consistent, and evenly-spaced peaks and troughs. It's known as a perfect or ideal wave because it has no irregularities; technically, a sound wave that looks like this is known as a *sine wave*. It's possible to create the sound of a sine wave using a synthesizer or audio test equipment. It's also possible to hear a close approximation of a sine wave with a tuning fork and even a flute. These are instruments which produce almost perfect vibrations.

Figure 1.6 shows a plot of the tuning fork's sound wave. When we plot a sound wave in this manner, we can call it a *waveform*. This term is used a lot, not only by

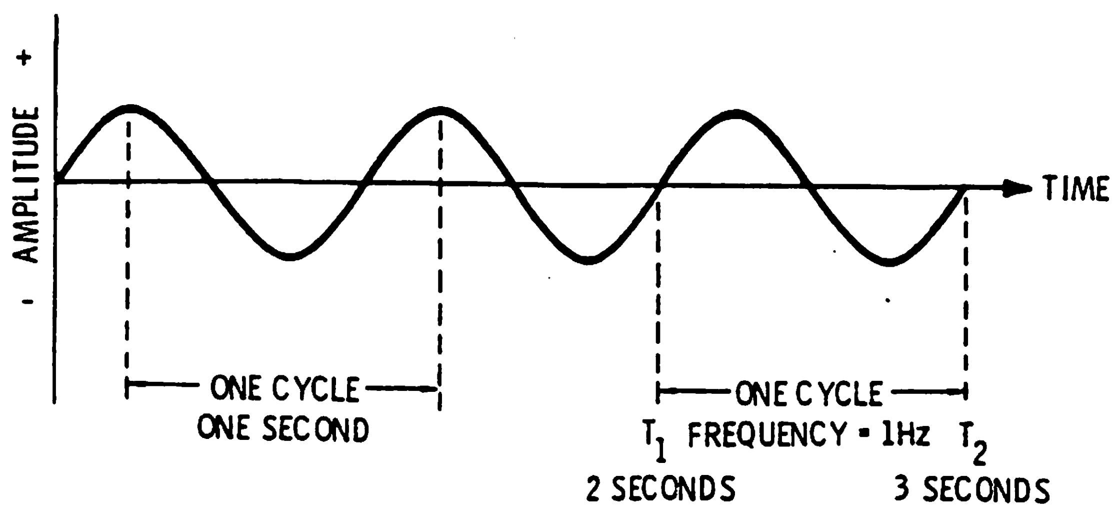
FREQUENCY

Frequency, in its simplest form, is a measure of how often, or "frequently" an event repeats itself. For a sound wave, frequency is the number of times that it passes through all of its negative and positive excursions and returns to its zero point. A sound source, such as a tuning fork, which vibrates back and forth 1,000 times per second is said to have a frequency of 1,000 cycles per second, or Hertz. The unit Hertz is named after Heinrich Hertz, a German pioneer in the transmission of radio waves.

Most sounds are complex and cannot be describes by a single frequency. The sound of a bell, for instance, contains very many frequencies at the same time.

Frequency of a signal in large part determine its PITCH. The more cycles per second, or Hertz, the higher the PITCH.

The range of human hearing is from approximately 20 Hertz to 20,000 Hertz (20 KHz). Human hearing is most sensitive at approximately 3,000 Hertz (3 KHz) to 4,000 Hertz (4 KHz). Middle A on the piano has its fundemental frequency at at 440 Hertz. Low A on the piano is 27 Hz,



ıg. 1-2. A cycle of a wave can be considered to begin at any point on a waveform.

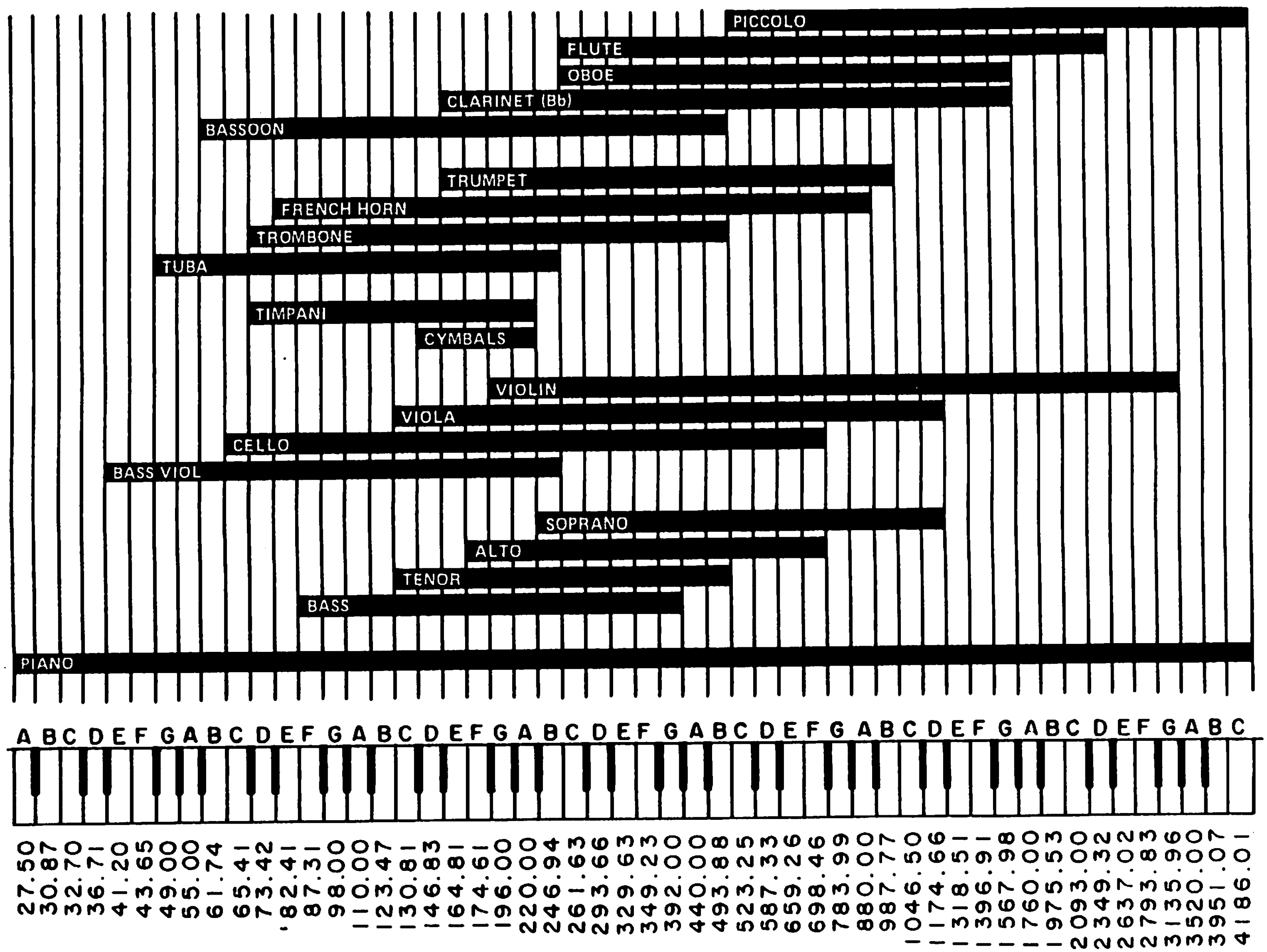
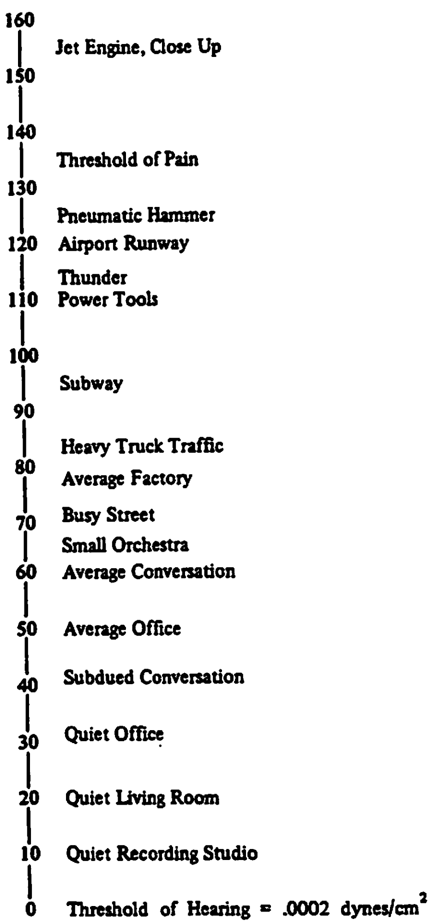


Figure 2-1. The frequency range of various musical instruments.

AMPLITUDE

The positive and negative excursions of the sound wave represent the increases and decreases in the atmospheric pressure of the air caused by the sound source. The distance above or below the zero line is called the AMPLITUDE and it measures the strength of a sound or signal. The amplitude of a soundwave is measured in decibels of sound pressure level. The human ear detects the amplitude of a sound wave as determining its VOLUME or LOUDNESS.

The minimum volume change that the human ear can detect is approximately 1 decibel (dB). The dynamic range of human hearing is defined as beginning at the threshold of hearing and ending at the threshold of pain, which encompasses approximately 140 db SPL (sound pressure level).



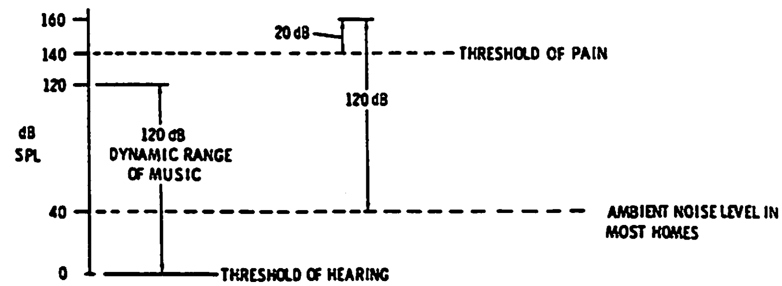


Figure 1-1. Typical Sound Pressure Levels

Radio Frequencies and Modulation

The Radio

The idea of transmitting an electrical signal on a wire means that the wire is the conductor, or the carrier of the electricity. There is a source (like a video camera) and a destination (a video monitor), and a means of connecting the two electrically (the wire).
A radio broadcast system adds two elements into the transmission system where wires normally would be: a transmitter to send the signal and a receiver to receive the signal. Television broadcast works identically to radio broadcast. Instead of using wires as the carrier of the signal, *radio waves* (electromagnetic energy in the range of radio frequencies) are used to carry the signal. Radio transmission systems are called wireless systems.

Radio Frequencies

Radio frequencies are a specific part of the electromagnetic system, just like light. In fact, the only difference between a radio wave and a light wave is that radio frequencies are lower than the light frequencies. Radio frequencies vibrate less times in one second than light frequencies. This is not to say they are not fast. Low end radio frequencies are tens of thousands of times per second. Imagine trying to wave your hand one hundred thousand times per second. You would be moving your hand at a typical radio frequency. Light vibrates trillions of times per second.

What's useful about Radio Frequencies?

Notice that you have never seen a radio frequency. This is simply due to the fact that *all you can see* is light frequencies. Any frequency out of the range of visible light you cannot see. That does not mean it's not there.
Light travels all the way from the sun to the earth, bounces off some objects and is absorbed by others. What the human eye detects as vision is all of these light waves bouncing off objects in the world.
Radio frequencies are also bouncing and absorbing off different objects in the world, but your eyes can't detect frequencies that low. Because the human eye can't detect it, it is hard to imagine they are there.
Imagine a flashlight pointed at a wall. The flashlight would be considered the source of information and the wall would be considered the destination. The information to be transmitted would be simple... on or off. When the flashlight is turned on, a spot on the wall lights up. Someone watching the wall would detect that the flashlight is on. When the flashlight is off, there would be no light on the wall, and that would be registered as the flashlight being off.
In this example, the flashlight is the source, the wall is the destination, and light is the carrier of information. Now imagine a "flash-radio" as opposed to a flashlight. When it is turned on, someone will detect that a radio wave is bouncing or passing through the wall. When it is turned off, someone will detect that there is no radio wave. Notice that someone *watching* won't see anything because their eyes won't register energy in the radio frequency domain. What's needed is an appropriate detector of radio waves.
What's so exciting about this is that instead of wires, radio waves can be used to transmit information.

How to transmit information
Imagine a rope lying flat on the ground. The flatness represents no alterations in height. It is uniformly flat. If you were to hold one end and move the rope with a whipping motion, a wave would move its way down the rope. Quickly, the wave would die out and the rope would again be flat. During that brief moment there would be information on the rope not normally present. Causing that information on the rope is called *modulation.*

Ways of modulating
In a television system, the information to be transmitted is brightness describing a scene. When a certain part of an image is bright, the rope (radio frequency) should have a high strength (amplitude). When a part of the image is dark, the radio frequency should be low (not much amplitude - the rope is near flat).
A video broadcast system requires two transductions: light energy is transduced to electrical energy (bright values - high voltage, dark values - low voltage) and then electrical energy is transduced to radio frequency energy (high voltage - high amplitude of the radio frequency carrier, low voltage - low amplitude).
This is process is called *amplitude modulation (AM).* High values of energy get transduced to high amplitudes of a radio wave and vice-versa.

Frequency Modulation
What is interesting to note is that there is an entire range of frequencies in the electromagnetic spectrum. And there is a range of ranges, like radio, light, x-rays, etc. One range is light frequencies. Lower frequencies tend to look red to the human eye, and higher frequencies tend to be blue and then violet. Past a certain frequency, the electromagnetic spectrum cannot be detected by the human eye.
Radio waves are an entire range of frequencies. A radio wave vibrating 100,000 times per second is different than one that vibrates 100,001 times per second. So for every frequency there is a unique, distinguishable wave.
Imagine the rope is now vibrating at a constant radio frequency of 100,000 time per second, or 100,000 Hz, or 100 kiloHertz (kHz). Because the frequency of the rope never changes, both the source (transmitter) and the destination (receiver) expect to see a vibration of 100 kHz. The consistent frequency means that the destination expecting to see 100 kHz may disregard it. In other words, imagine if you lived your entire life on a ship on sea. You would quickly become accustomed to the rocking of the ship and eventually you would not notice it was there. If you jumped up in the air and landed back on the ship's deck, you would notice this sudden and unusual displacement, but you would still not notice that the ship is rocking. This is because the rocking of the ship is inherent to the system. If you suddenly took shore leave on land you would notice that the environment changed, and the rocking of the ship would be missing.

70

A radio wave constantly vibrating at 100 kHz carries no information, like a person living on a ship. Nothing is unusual about this signal. The transmitter sends a frequency of 100,000 vibrations per second and the receiving device (called a receiver) is *tuned* to receive a radio frequency of 100,000 vibrations per second. Both points of the system are now in agreement about what the transmitting media is: a radio frequency of 100,000 vibrations per second. But there is no information contained within that media. To modulate the carrier with useful information, you must take, for example an electrical signal, and when the voltage is high, increase the amplitude of the 100,000 Hz radio frequency. When the voltage is low, decrease the amplitude of 100,000 Hz frequency. In both cases, the frequency stays the same, but the amplitude changes. This, again, is amplitude modulation of a radio frequency carrier.

Frequency modulation (FM) is exactly what the name implies: applying information to a carrier frequency (for example, 100,000 Hz) by modulating, or changing, the frequency proportional to the information signal.
Here is how it works: an electrical signal with a high voltage (representing brightness in a picture) will increase amplitude of a 100,000 Hz carrier frequency when amplitude modulation is used. In frequency modulation, the amplitude remains constant, but high electrical voltages representing brightness in the picture will increase the frequency of the carrier signal. Therefore, a high electrical voltage will frequency modulate the carrier wave of 100,000 Hz to 101,000 Hz. This means the frequency has become higher. The receiver will detect a higher frequency and translate that to mean a high electrical voltage. A low electrical signal at the transmitter will frequency modulate the carrier wave of 100,000 Hz to a lower frequency, perhaps 99,000 Hz. The receiver will detect a frequency lower than 100,000 Hz and translate that to mean a low electrical signal.

In both amplitude modulation and frequency modulation, a carrier wave is used. This is the environment in which the information will be passed. In a system with wires, the wire is the carrier. In radio systems, specific radio frequencies are the carriers. Amplitude modulation uses electrical voltages to increase or decrease the strength (amplitude) of the carrier frequency. Frequency modulation uses electrical voltages to increase or decrease the carrier frequency's original frequency. In both cases, the carrier frequency can be disregarded or eliminated at the receiving end, and what is left is the varying strengths of the original electrical signal.

Electricity

Possibly the most important discovery of the last two centuries, electricity is ever present in the modern world. Electricity runs street lights, radios, TVs, computers, hair dryers, clocks, air conditioners, light bulbs... and yet electricity itself is never seen. Although human life has been greatly convenienced by this discovery, almost no one knows how it works or what it is.

How to go about learning about electricity

Ideally, all students of video will take a course in basic electronics to understand the principles upon which the video industry was built. This section goes over a basic explanation of the fundamental nature of electricity.

It is important to recognize that electricity was not invented but discovered. Once this discovery was made, in the late eighteenth century, inventions based on electricity exploded exponentially. Still, one question remains...

Okay, what is electricity?

Determining *what* electricity may be less practical than understanding the effects electricity has on different materials and vice-versa. For example, you may already have an intuition about a rule that metal objects conduct electricity. This is why wires are made of metal instead of leather, paper or rubber.

How does it happen - electricity?

Electrical occurrences happen in nature. Lightning is one example of this. Static electricity caused by scuffing your feet across a carpet is another example. What happens to cause the electricity?

Electricity occurs because all matter is electrical. Every atom in the universe has some electrical charge. This means electricity could occur anywhere, potentially. The transfer of electrical energy from one object to another is what we call electricity. When there is no noticeable transfer of electrical energy, we do not notice electricity present. Therefore, a large transfer of electrical energy from the ground to the clouds might be called lightning (electricity). A shock you receive from touching a metal doorknob is called electricity because there is a transfer of electrical energy from your finger to the doorknob.

The transfer of electrical energy, which is present in all matter, is called electricity.

How to tame electricity... Atomic structure

Controlling electricity is a simple matter of understanding how it behaves. The best way to study this is to cause electricity (the transfer of electrical energy from one object to another) to happen. The discovery of electricity meant human beings could now cause electricity to occur instead of waiting for a thunder-and-lightning storm to strike.

Understanding the nature of matter also helps to control electricity. According to modern science, atoms are made up of several subatomic particles: protons, electrons and neutrons. As their names imply, protons are positively charged, neutrons are neutrally charged (charged neither positive or negative) and electrons are negatively charged. This means that within a single atom, protons and electrons can balance out their opposing charges.

The world-famous electron

Electrons, very tiny subatomic particles, are the fundamental measure of electrical force. They are considered to be negatively charged. Every atom has some amount of electrons... from zero to over one hundred. Consequently, a large group of atoms also has a number of electrons. The more atoms in an object, the more electrons.

An atom which is atomically and electrically neutral has a charge balanced by an equal number of electrons and protons.

An atom which is negatively charged has an excess of electrons, just waiting to jump off to a receiving atom.

An atom which is positively charged has a lack of electrons, just waiting to receive electrons.

The same is true of larger objects. An entire object which has an excess of electrons is considered negatively charged because their are more electrons (negative charges) than protons (positive charges). An object with an overall lack of electrons to balance out the number of protons is considered positively charged. Don't be confused by the inverse relationship. Excess electrons means negative charge. Lack of electrons means positive charge.

It would help to see all of this

For all practical purposes, you cannot see electricity. This fact usually keeps people from delving very far into discovering how it behaves. Generally, people imagine electricity as bright, crackling energy that shoots from place to place. If that were true, our world would be ablaze with flashing, sizzling energy transfers. Notice that although electricity is probably flowing in the very walls in your house, you do not even notice it is there. That is the relationship between humans and electricity, which explains why it was only discovered very recently. It is a subtle fundamental of nature. But don't be deterred! What you can't see exists whether you see it or not. Accepting electricity as a fundamental of the universe and knowing the rules under which it behaves is the only route to fully understanding electrically based inventions, from toasters to video cameras. Start to note everything that runs on electricity in your house, on your street, at your job. All of these things are electrons flowing from one place to another.

But what is an electron? That is a good question worth asking. Just don't expect an immediate response from anyone. Electrons are related to electricity and they are a part of atomic structure, but that doesn't say what one is, free standing in the universe. Even today, no one can fully define what a electron is, or where it came from.

Electricity flows

The way electricity "flows", in other words, the way electrical energy is transferred from one object to another, is via a potential electrical situation.

Imagine an empty glass on a flat countertop. The glass has space to contain something, like water. The countertop will not contain water very well. Consider the glass to be positively charged. The countertop is neutrally charged. Now imagine a pitcher of water poured from above the countertop and the glass. The pitcher is negatively charged. The water hitting the countertop will bounce off or fall on the floor. The water hitting the glass will fall into it and fill it up until the water begins to spill out onto the countertop and onto the floor.

In this example, the transfer of electrical energy happened between the pitcher which contained water (excess electrons - negatively charged), and the empty glass (lack of electrons - positively charged). Note that the countertop (balance of protons and electrons - neutrally charged) did not accept the water (excess electrons) from the pitcher.

What this indicates is electricity occurs whenever there are two conditions: an object has an excess of electrons and a second object has a lack of electrons. This creates a situation in which the excess electrons tend to flow towards the object where there is space awaiting them. The presence of both a negatively charged object and a positively charged object means their is the *potential* for electricity (electrical energy to be transferred from the negative object to the positive object).

Potential implies that electricity *could* flow, but that it doesn't necessarily. This potential, however, is inherent in any electrical flow. Without the potential for the situation to occur, it will never occur.

How electricity gets from point A to point B

Electricity flows from a negatively charged object to a positively charged object. This means an excess of electrons flows to fill the space where electrons are lacking. But how do the electrons get there?

A conductor of electricity is a material which facilitates the flow, or movement of electrons. Metals are particularly efficient at allowing electrons to move through their molecular structure. When there is a negatively charged object *at one end* of a metal wire, for example, and a positively charged object *at the other end* of a metal wire, there is a potential electrical difference. The metal wire makes the potential electrical flow actually occur.

Voltage! What does it mean?
Voltage is simply the potential electrical difference between two objects. If two objects are electrically neutral, there is no potential difference between them, so there is no voltage. However, if one object has an excess of electrons (negative charge) and the other has a lack of electrons (positive charge), there is a potential difference between the two. This potential difference is quantified with the measurement unit *volt.*

Scenarios causing voltage
Positive charged object (lacking 5 electrons) and negatively charged object (excess of 5 electrons) - produces voltage equal to the amount of positive charge plus amount of negative charge (5+5=10 electrons difference)

Neutrally charged object (balanced number of electrons) and negatively charged object (excess of 5 electrons) - produces voltage equal to the amount of negative charge (0-5 = 5 electrons difference)

Positive charged object (lacking 5 electrons) and neutrally charged object (balanced number of electrons) - produces voltage equal to the positive charge (5+0 = 5 electrons difference)

Neutrally charged object (balanced number of electrons) and neutrally charged object (balanced number of electrons) - produces no voltage (0+0 = 0 electrons difference)

Notice that voltage exists only when their are two objects to compare charges. One object alone does not generate voltage. In fact, one object relative to a second object has one voltage, but relative to a third object it may have a different voltage.

The greater the excess or lack of electrons is, the greater the potential electrical flow will be, because their is more of an imbalance in the system. The greater the imbalance means a greater potential of electrical flow. A greater potential of electrical flow is a greater voltage. High voltage represents a stronger imbalance and a stronger push to get electrons flowing. Low voltage represents a smaller imbalance and a smaller push to get electrons flowing.

Pushing electrons
Electricity is nothing but pushing around electrons. To cause electricity, all it takes is creating a situation with a negatively charged object and a positively charged object. In other words, creating a voltage. A similar scenario might be building a dam which collects millions of gallons of water ready to flow if only the dam weren't blocking it. There is the potential of a large flow of water, not yet realized. Adding a pipeline in the dam will allow water to flow. The pipeline is the conductor, the millions of gallons of water at one end versus zero gallons of water at the other end is the voltage.

Amps! Current flows
The smaller a pipeline is, the less water that will flow through during each second of time. The amount of water passing in a pipeline per second is the measure of current.

How much water passes a point every second? Measuring this will give you the value of the current. In an electrical system the same holds true for electrons. The number of electrons passing a point in one second is the current. The unit of measurement for the number of electrons passing a point in one second - current - is called *amperes* or *amps*.

Voltage and current, hand in hand
Voltage describes the strength of the push that electrons will receive, caused by the potential electrical difference between two objects. Current describes the amount of electrons that pass a point in one second. The stronger the voltage, the higher the number of electrons that pass a point in one second. Therefore, higher voltage causes higher current and lower voltage causes lower current. The width of the conductor (pipeline) also determines how many electrons will pass in one second. The wider the path, the more electrons will be able to pass in one second.

Resistance: something in the way
Different materials make electron flow more or less efficient. Materials with a low efficiency for electron flow have a high *resistance* to electrical flow. Objects like rubber and plastic have such high resistances to electrical flow (due to their molecular structure) that they are not called conductors but insulators. They inhibit the flow of electrons. Likewise, materials with a high efficiency for electron flow have a low resistance. These materials are called conductors.

Resistance is measured in units called Ohms (W).

All materials which allow electron flow have some resistance. So there is resistance to every push of electrons. In other words, while the voltage is pushing electrons, there is resistance pushing back on the electrons. The actual current in any electrical system is a result of the voltage push and the resistance reactionary push.

Ohm's Law
Ohm's law mathematically states the relationship between current, voltage and resistance.

Voltage = Current x Resistance
Volts = amps x ohms (W)
V = I x R (electrical terminology)

Batteries

Batteries are simply containers of electrical potential. There is a positive terminal and a negative terminal. In other words, one side has an excess of electrons and the other has a lack of electrons. Both sides would easily balance each other out if only they were connected. The trick with batteries is keeping the two terminals isolated from each other yet contained in the same package. This is done with an insulator which has such high resistance it does not allow electrons to flow between the two poles within the battery. However, if the two terminals are connected via a conductor (a wire), the potential between the two poles is instantly realized. There will be a flow of electricity until both sides have balanced and neutralized each other. Beware of doing this! The electron flow between the positive and negative poles without any resistance can be too fast and the battery may overheat and possibly melt.

Batteries contain two materials: one that has had electrons removed from it and another material with excess electrons added to it. Note that different types of batteries are distinguished by the materials within them: Nickel-Cadmium (NiCad), Lead-Acid, Nickel-Hydride, etc.

Because the flow of electricity always happens at one particular potential or voltage, this source of electrical energy is called *direct current(DC)* electricity.

Alternating Current

A system with a changing voltage means the electrical potential between two objects changes over time. It is not a consistent voltage over time like a direct current system. The only difference between an alternating current and a direct current is that the voltage (potential) is constantly fluctuating in AC and is consistent in DC.

Specifically, alternating current fluctuates from a positive voltage (flow from object A to object B), to a neutral voltage (no flow between A and B), to a negative voltage (flow from object B to object A) and then back again. This constant alternating of the current gives the name AC. Note that the same fundamental rule of electricity applies: a potential, voltage, or lack and excess of electrons at two points is required for any electrical flow to happen.

The outlets in your house

The electricity in your house is delivered from power plants via wires carried on telephone poles. The electricity carried to buildings is alternating current (AC). This is because AC can travel longer distances than DC can.

How fast does AC alternate?

The speed that AC fluctuates from positive voltage to neutral voltage to negative voltage back to neutral and finally positive voltage again is called the *frequency* of the alternating current. All electrical equipment that runs using AC is built to work with a specific frequency of alternating current. If every device required its own special frequency of fluctuating alternating current, it would be impossible to decide what frequency of AC to send to outlets in buildings.

The solution is to standardize the way AC electricity will behave. Alternating current is a specific application of electricity designed by humans. Therefore, how it behaves is controllable. The frequency can be changed and so can the limits of positive and negative voltage.

In the United States, AC electricity alternates 60 times every second or 60 Hz. It's peak voltage is 120 volts positive and 120 volts negative. The average voltage over time is between 110 and 117 volts. This means that every second, the voltage alternates 60 times between positive, neutral, and negative values.

In other countries in the world, there are different AC electrical standards. In Britain, AC frequency is 50 Hz, and the maximum voltage is 240 volts. This means a device built for use with an American standard AC will most likely be overwhelmed by British voltage (strength of push of the electrons) which is twice as much as it is expecting. It will fry your equipment, in other words. Thus, you buy an adapter.

Impedance - something in the way

Resistance is deters the flow of electrons in a direct current system, where there is only one voltage.

Impedance deters the flow of electrons in an alternating current system, where there are many different voltages possible. The impedance on an alternating current system is based on the frequency (how fast) the current is alternating. The higher the frequency, the more impedance is present. This is a rule of behavior of alternating current systems.

AC versus DC

Most electrical devices run on specific, direct currents. This means things like video cameras, computers, flashlights, all operate with DC.

Type of Electricity	Advantages	Disadvantages
Direct Current	Easy to work with, most consumer products work with a consistent, steady voltage (DC)	Does not transmit far before being overwhelmed by resistance
Alternating Current	Long transmission distances possible, changing voltages can represent changing information, easy to amplify	Complicated to work with, impedance (AC resistance) changes depending on frequency of AC, most devices work with a consistent voltage

How can you plug a video camera in a wall outlet if the video camera is DC-based and the outlet is AC-based? Or a computer... or a stereo system? You need an AC-to-DC adapter.

Transduction

Transduction is the process of proportionally turning one form of energy into another. The entire energy of a system is always conserved, but the form of the energy may change.

Ear: sound energy (pressurized air molecules) - electrochemical energy (brain activity)
Eye: light energy (electromagnetic vibrations in visible light range) - electrochemical (brain activity)
Microphone: sound energy (pressurized air molecules) - electrical energy
Speaker: electrical energy - sound energy (pressurized air molecules)
Video tube: light energy (electromagnetic vibrations in visible light range) - electrical energy
Cathode ray tube (CRT): electrical energy - light energy (electromagnetic vibrations in visible light range)
Light bulb: electrical energy - light energy (electromagnetic vibrations in visible light range)
Motor: electrical energy - kinetic, physical energy
Generator: kinetic, physical energy - electrical energy

Why transduce energy?
Representing one form energy as another form of energy allows control over energy otherwise not possible. Although the medium (energy type) has changed, the information represented has not.

Transducing sound energy in the air to electrical energy, all innovations in electrical technology are now applicable to the sound energy. The implications of this are vast. Light energy and sound energy can both be transduced to electrical energy. At this point, there is a myriad of possibilities. The signal can be increased or reduced, or specific frequencies can be increased or reduced. Parts of the signal can be eliminated or processed while others left untouched. This means the original information can be changed in the electrical realm, using knobs and transistors instead of filters and lenses (in the case of light).

No one can hear or see electricity
The most important thing to remember is that keeping information in the form of electrical energy is almost completely useless. At some point, it must be transduced *back* into its original energy form. This gives the appearance that the original energy form was altered, though this is not so. It was the *information* contained within the original energy form that was altered (while in the form of electrical energy).

Transduction - the two steps
For transduction to have practical use, two steps are necessary. In the case of light energy to be processed, the two transduction steps would be: light energy transduced to electrical energy, then electrical processing, and then the electrical energy transduced back to light energy. The human eye can see the light before the transduction and after, but cannot interpret the information while in its electrical energy form.

Transduction, and the types of energy we transduce, are completely based on the human senses. This is ultimately the end result... energy forms which can be detected by human audiences.

Time

Breaking down time

Time is yet another force or dimension of nature which cannot be observed directly. Like electricity and light, humans don't see the actual forces as much as the effects of those forces. The effect of light on human eyes is vision. The effect of electrical energy between two electrically different objects is a flow of electricity, which sometimes can be seen as light in lightning or light bulbs. The effect of time might be considered by humans to be "cause" and "effect", or "before" and "after".

Time appears to move constantly, relentlessly. The sun is up in the sky and at another moment it is down past the horizon. The difference between these two moments can be measured in time. While running, one foot is off the ground in one moment and in another moment the foot is on the ground. What happened that got the foot from the air to the ground? If time stopped, there would only be one state of the universe. The movement of time is measured in shifts of the state of the universe.

If you wave your hand in front of your face, you will notice that it appears to blur. This is because human vision can only detect changes in light information at a certain speed. Past this speed, things begin to blur and discrete locations of objects are impossible to detect. In other words, if you slowly move your hand back in forth in front of your eyes, you will be able to see your hand in a number of discrete positions. As you speed up the waving motion, you can still see discrete positions until the rate of waving is high enough that your hand appears to blur. Your hand is still in discrete positions in space but it is moving too fast for *your eyes* to detect.

Slowing down time

Generally speaking, time is moving consistently. This means it always takes 24 hours for the earth to revolve around its axis, and 365 days for the earth to orbit the sun.

24 hours = 1 rotation of the earth around its axis
1440 minutes = 24 hours x 60 minutes per hour = 1 rotation of the earth
86,400 seconds = 1440 minutes x 60 seconds per minute = 1 rotation of the earth
86,400,000 milliseconds = 86,400 seconds x 1000 milliseconds per second
86,400,000,000 microseconds = 86,400,000 milliseconds x 1000 microseconds per millisecond

Therefore, 1 rotation of the earth around its axis occurs in the duration of 1 day, 24 hours, or 86,400,000,000 microseconds. Generally, humans don't notice events occurring during 1 microsecond (1 millionth of a second). It takes longer amounts of time, like one 24th of a second, for a human to detect the passing of time.

Time can be continually divided into smaller and smaller intervals:
1 day = 24 hours
24 hours = 60 minutes
1 minute = 60 seconds
1 second = 1000 milliseconds (milli- = thousandth)
1 second = 1,000,000 microseconds (micro- = millionth)
1 second = 1,000,000,000 nanoseconds (nano- = billionth)
1 second = 1,000,000,000,000 picoseconds (pico- = trillionth)

It is hard to imagine 1 trillion discrete events occurring in one second. If it were possible to perform tasks at the picosecond level, just think what you could get accomplished during the day. Watch a clock tick or flash as the seconds go by. Notice how long it feels. That perception of time is how humans gage things happening. If you can't imagine things happening any faster than one second, it is hard to comprehend the notion of an event that only lasts one microsecond, let alone nanosecond or picosecond.

The VERY BIG and the very small

The exploration of the fundamental nature of the universe leads to events occurring in magnitudes not normally perceived by human beings. When attempting to understand light, radio and sound energy for what they are, a leap in perception is necessary. Because eyes can't be altered to perceive things faster or smaller, the only thing to be done is to formulate a model about how things are happening even though you can't really perceive them. For now, it is reasonable to accept that phenomena like light waves are vibrating trillions of times in one second and that the length of one vibration of light is very, very small (millionths of one meter). You will never see these tiny intervals of time or distance, but they do exist.

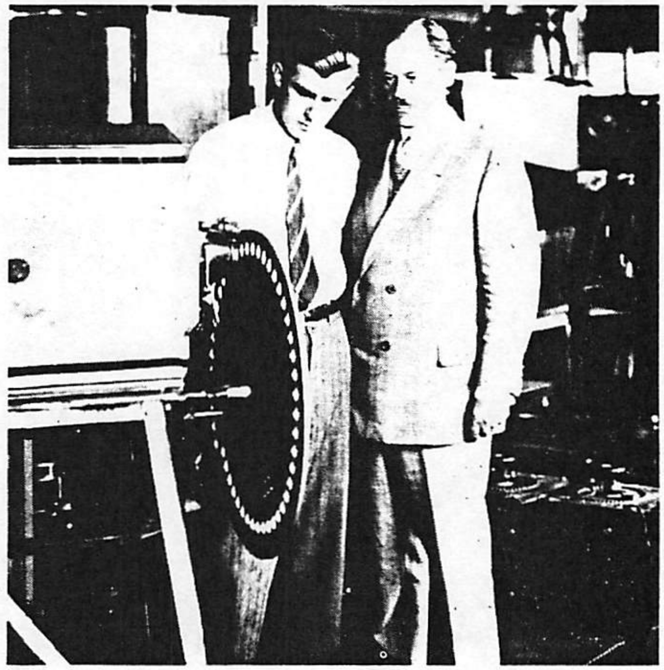
THE HISTORY OF TV TECHNOLOGY

JOHN P. TAYLOR

Television was not invented by any one man. Nor did it spring into being overnight. It evolved gradually, over a long period, from the ideas of many men—each one building on the work of his predecessors. The process began in 1873, when it was accidentally discovered that the electrical resistance of the element selenium varied in proportion to the intensity of the light shining on it. Scientists quickly recognized that this provided a way of transforming light variations into electrical signals. Almost immediately a number of schemes were proposed for sending pictures by wire (it was, of course, before radio).

Early Systems

One of the earliest of these schemes was patterned on the human eye. Suggested by G. R. Carey in 1875, it envisioned a mosaic of selenium cells on which the picture to be transmitted would be focused by a lens system. At the receiving end there would be a similarly arranged mosaic made up of electric lights. Each selenium cell would be connected by an individual wire to the similarly placed light in the receiving mosaic. Light falling on the selenium cell would cause the associated electric light to shine in proportion. Thus the mosaic of lights would reproduce the original picture. Had the necessary amplifiers, and the right kind of lights, been available this system would have worked. But it also would have required an impractical number of connecting wires. Carey recognized this and in a second scheme proposed to "scan" the cells—transmitting the signal from each cell to its associated light, in turn, over a single wire. If this were



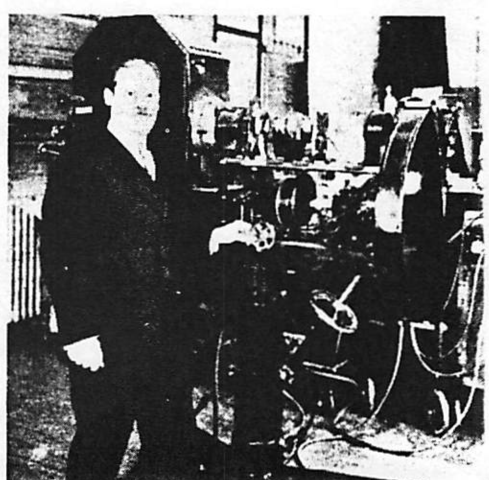
Ray D. Kell and Dr. E. F. W. Alexanderson, early television pioneers, pose with the mechanical scanning disk.

done fast enough, the retentive power of the eye would cause the resultant images to be seen as a complete picture.

The next forward step was to eliminate the selenium mosaic by scanning the image directly. In 1881 Shelford Bidwell, in England, demonstrated one way this could be done. He mounted a single selenium cell in a box with a pinhole. A motor-driven cam moved the box rapidly up and down and across the plane of the image. The system worked but the scanning speed

1928. E.F.W. Alexanderson, chief television consultant at General Electric, demonstrates the first home TV set. It has a three-by-four-inch screen; the control in Alexanderson's hand keeps the mechanical receiving apparatus in pace with the signal.

Alexanderson poses with his television receiver, an intricate device for projecting images on a screen from a position backstage. GE's station WGY in Schenectady, New York, begins the first regular programming, three times a week, on May 10, 1928.



FELIX THE CAT: STAR OF EARLY TELEVISION

The first "star" of television was Felix the Cat, a well-known cartoon character of the twenties. He made his debut before the TV cameras in 1929. RCA engineers needed someone to focus their camera on while they experimented with their equipment. They soon ran out of human volunteers—no one could stand the lights and the tedium for very long. So they mounted a foot-high statuette of Felix on a turntable and placed that in front of their camera. There he went round and round, hour after hour, day after day—and, in fact, year after year. What began as a temporary expedient became a fixture. Felix's career spanned most of the 1930's. His smiling visage became not only a symbol of early television, but also a sort of test pattern by which the engineers measured their progress. The pictures below illustrate this. At Felix's debut in 1929, the RCA engineers were using a 60-line scanning speed and the image appears as if seen through a venetian blind. The 120-line picture of 1932 is much better, although still very fuzzy. The 441-line picture—typical of television as introduced at the 1939 World's Fair—is quite sharp. These, of course, are just three of the steps in television's progress. There were many in-between steps—and Felix appeared in most of them.

was very limited. Soon afterward Maurice LeBlanc, in France, suggested that the image could be scanned by an oscillating mirror which would reflect the light variations onto a fixed selenium cell.

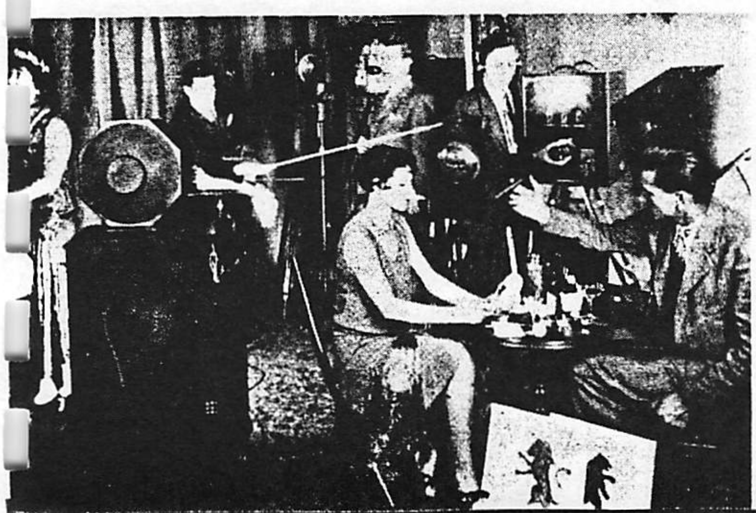
The Nipkow Disk

A number of other mechanical scanning arrangements were proposed at about this time. But it remained for a German, Paul Nipkow, to devise the simplest and most workable one. In 1884 he filed a patent on a system which used a scanning disk that had small holes spaced around the outer circumference in a spiral pattern. This disk rotated at high speed in front of the selenium cell at the sending end. As it did so, each hole scanned a line across the image to be transmitted. Because of the spiral pattern each scan line was slightly below the preceding line—until the whole image, top to bottom, had been scanned. An identical disk—rotating in synchronization—was placed in front of the light source at the receiving end. This produced lines of varying light intensity which reproduced the original image.

Nipkow never had the money to build a working model of his concept. But many who came after him did, and the "Nipkow Disk"—or a variation of it—was used in most television systems until the early thirties.

Electronic Experimentation

There were some early experimenters who were not satisfied with mechanical scanning. The cathode-ray tube had been developed by K. F. Braun in 1897, and soon after there were attempts to adapt it for television. In 1902 Boris Rosing, a Russian, started experimenting with cathode-ray tubes for image reproduction. In 1907 he described and actually built a system in which the picture was viewed on a cathode-ray tube in which the beam was deflected in synchronization with the sending signal. Rosing was a professor at the Technological Institute of Saint Petersburg, and one of his greatest legacies, perhaps, is that among his young pupils was Vladimir K. Zworykin.



September 11, 1928. WGY Schenectady, broadcasts the first TV drama, *The Queen's Messenger*. There is an individual camera for each performer and also one for props being manipulated by the two people at the table. The actors dress for their parts, although the small picture can show only their faces.

NBC has its first TV personality. Felix the Cat is transmitted from New York to Kansas during a series of 1928 experiments. His image is picked up on primitive 60-line viewers. Eventually, sophisticated viewers will have 525 lines.

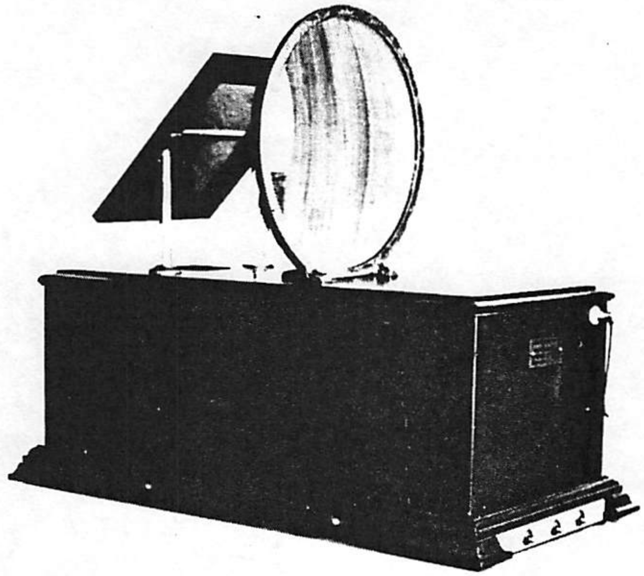
"Television was not invented by any one man."

Others were working along the same lines. In Germany a system using a cathode ray tube at the receiving end (much like Rosing's) was described by Dieckman in 1906. Neither he nor Rosing knew of the other's work. In 1918 A. A. Campbell Swinton, in England, suggested a system which would use cathode-ray tubes at both sending and receiving ends. He never built such a system but he is usually credited with being the first to suggest the idea of all-electronic television. In 1911 A. Sinding-Larsen suggested the use of radio as a carrier for picture signals. Thus, well before World War I, all of the concepts of what we think of as "modern television" had been put forward. But the onset of the war interrupted the early experiments, and for almost a decade television was forgotten.

Public Demonstrations Begin

In the early twenties there was a new burst of television activity. C. F. Jenkins, J. L. Baird, Zworykin, Philo T. Farnsworth and others were busy in their laboratories. And then there began to be public demonstrations. In 1925 Jenkins, in the United States, and Baird, in England, publicly demonstrated working (albeit crude) television systems using the Nipkow disk. Other experimenters were working behind locked doors.

This activity inevitably drew the attention of the big corporations. And with that came a marked change. The first fifty years of television development (from 1875 to 1925) had seen lonely inventors working by themselves or in small laboratories. But starting



The image in this Jenkins radiovisor is reflected by the mirror onto the magnifying glass where it is viewed.

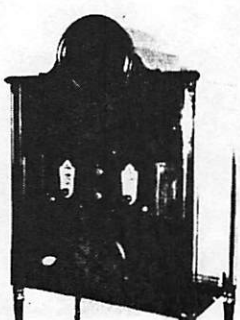
about 1925 the big radio companies undertook major television development programs. Public demonstrations of operating television systems were made by AT&T (1927), General Electric (1928), RCA (1930), Dumont (1930), and Philco (1931).

All of these systems used mechanical scanning systems employing either the Nipkow disk or something similar. The very earliest were 30-line systems—that is, the scene to be transmitted was scanned by 30 horizontal lines. Later systems increased this to 60 lines, to 120 lines and, in one case, to 243 lines. But even then the pictures were not sharp and the large disks required (some four feet in diameter) were obviously not practical for home use.

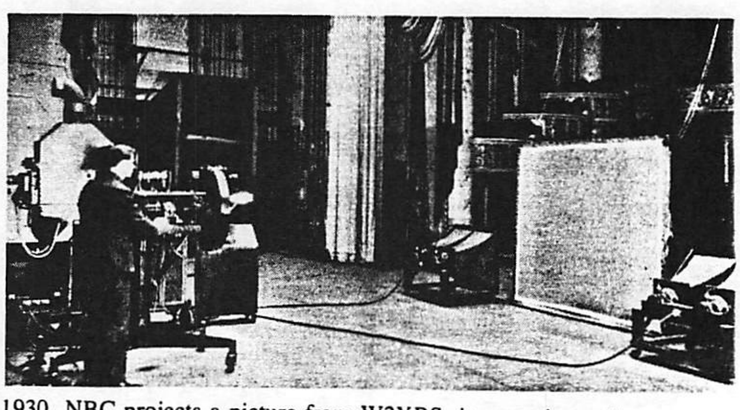
Beginning of the Electronic Age

Fortunately, a better system was on the horizon. In 1919, Zworykin had emigrated to the United States

1929. Now working for RCA, Vladimir Zworykin demonstrates the kinescope, a cathode-ray TV receiver, which along with the iconoscope completes an all-electronic TV system.



C. Francis Jenkins's receiver with a round screen.



1930. NBC projects a picture from W2XBS, its experimental TV station in New York, onto a six-by-eight-foot screen at a midtown theater.

"The process began in 1873..."

and gone to work for the Westinghouse Company where he continued the study of electronic television which he had begun (under Professor Rosing) some ten years before. In 1923 he filed his first patent on the iconoscope, the first camera pickup tube. In the iconoscope the picture to be transmitted was focused on a photosensitive mosaic mounted in the tube. The mosaic was scanned, line-for-line, by an electron beam which released a small electric charge from each tiny photoelement as the beam passed over it. Thus the basic concept of scanning was continued. But now there was an important difference. The speed of the beam and the number of lines scanned were no longer limited by mechanical considerations.

In November 1929, Zworykin demonstrated the kinescope—a greatly improved cathode-ray television picture tube. Now, with an iconoscope at the sending end and a kinescope at the receiving end, an all-electronic system was a practical possibility.

RCA Backs Electronic Television

Zworykin took his ideas for developing such a system to David Sarnoff, then vice-president and general manager of RCA. Sarnoff was enthralled. Within months Zworykin transferred to RCA where he was made director of electronic research. He was provided with the money and manpower he needed to carry out his project. Equally important, other groups within RCA engineering and research were set up to develop needed system elements (cameras, transmitters, receivers). RCA had decided to place all its bets on electronic television.

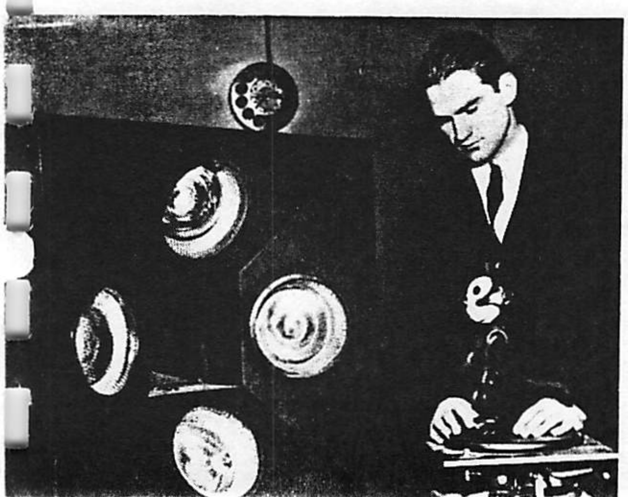
During all of the 1930's RCA (and other companies, too) worked continuously to improve television quality. The measure of their progress is indicated by the increasing number of scanning lines employed in the successive demonstrations which RCA made during this period:

DATE OF DEMONSTRATION	NO. OF PICTURE LINES
1930	60 lines
1931	120 lines
1933	240 lines
1936	343 lines
1939	441 lines
1941	525 lines

In 1939, RCA announced that coincident with the opening of the New York World's Fair it would start regular television broadcasting, and would offer receivers for sale to the public. It did—but not for long. Other manufacturers complained and the FCC rescinded its approval for commercial television while an industry committee reviewed the standards. In the end, this committee recommended the basic RCA system, but changed the number of lines from 441 to 525. The FCC then approved the start of commercial television. RCA quickly changed its equipment to 525 lines and went back on the air. But within months came Pearl Harbor, and television once again went into mothballs.

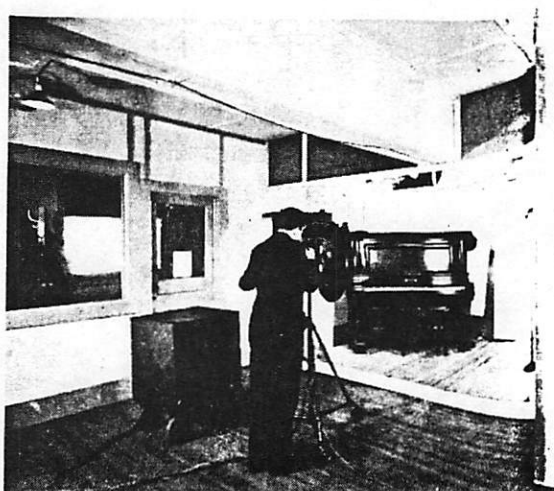
The Postwar Boom

Shortly after World War II ended RCA introduced a new television receiver—the famous "630"—which was a tremendous advance over prewar receivers. And, to stimulate production, it gave other companies the manufacturing drawings. The market exploded. New television stations came on the air at a fast pace. Some interference problems cropped up and the FCC



Once again, a model of Felix the Cat is used in an early broadcast demonstration at W2XBS. For hours, Felix revolves on a phonograph turntable while tests are made. Felix is used because humans cannot tolerate the hot TV lights for long periods.

1931. A program promoting home TV is broadcast from the Dumont station WXCD in Passaic, New Jersey. In 1938, Dumont will place the first all-electronic home television set on the market.



Video

Why video?
Video allows communication of light information between two points, or two people.

What is video?
Video is light converted to electricity converted to light in real, discrete intervals of time.

What are the advantages?
Immediacy: light is almost immediately transduced to electricity and then back to light, giving instant processing power in the electrical domain.
Technology: as electrical technology advances, all video processing capabilities advances
Transmission: electrical information can be transduced to radio energy and broadcast over a large geographical range
Storage: electrical information can be transduced to magnetic energy and impressed upon inexpensive magnetic tapes for permanent storage of an original image of light

Let's build an eyeball
What does an eyeball do? It takes light energy and transduces it to electrochemical nerve impulses which are sent to the brain. Building a light-to-electrochemical transducer requires a huge study of biological and neurological facts. For the sake of efficiency, we will keep the invention of the eyeball as it is and invent another device which takes light energy, processes it, then turns it back to light energy, which the eye can then process and interpret.

Video is a poor mimic of human vision
All that a video system does is mimic the functions of the eye-brain/light-nerve impulse transduction. Really, it is not a very accurate mimicry. This is why watching a scene on a TV screen is a completely different experience than actually *being there*. The same is true of watching a film projection. The resolution of the eye which results from tiny photochemical receptors sending information to the brain is not present in modern video systems. The way to the humans sense of vision is still through the eye, not directly to the brain.

How does the video transducer work?
Several light-to-electricity transducers have been invented. The first was called a *video tube*. It was a glass tube with all matter evacuated from it, thus creating a vacuum (lack of all matter) within. The primary element of the transducer is the *photoelectric* coating on the surface of the tube. Photoelectric materials receive photons (particles of light energy) and give off a proportional number of electrons (particles of electrical energy). Other photoelectric materials, known as *phosphorescent* materials, work in reverse. Phosphors give off photons (light energy) when electrons are received.

Photons are received, electrons are released. This is the process of transduction.

The back of the video tube is at a constant voltage (electrical potential). The front surface of the video tube (transducer) absorbs protons and produces electrons. When photons hit a point on the surface of the tube, a proportional amount of electrons is produced, which means there is an excess of electrons at that point on the surface of the tube. The electrical potential between the point on the surface of the tube and the back of the tube is the voltage of the video signal for that moment of time. If the image was bright at that point, a high number of protons was radiated and absorbed by that point on the surface of the tube. That high number of protons is transduced to a high number of electrons, which means a high electrical potential (voltage) compared to the constant electrical potential (voltage) at the back of the tube. If the point of the image was dark, few photons would reach the tube surface and few electrons would be emitted. This would be a low voltage between the tube surface and the back of the tube. No matter what the brightness is of an image at a given point, the number of photons will be translated into a corresponding number of electrons, which will create a corresponding voltage, or strength in which to push the electrical video signal.

The challenge of inventing video

Imagine a framed scene. The scene is composed of varying brightness values. Bright points of the scene are caused by high amounts of radiated light energy. Dark points are caused by low amounts of radiated light energy. In one moment of time, a scene is a whole series of bright and dark points radiating light energy.

Now consider electrical energy. At any one moment of time, electricity can be a high voltage (potential electrical difference between two points) or a low voltage (little potential difference between two points). But one electrical signal cannot be high or low at the same time!

The challenge is immense: how can many varying levels of light in an image, occurring in one instance, be transduced onto an electrical signal which can only be one level at one moment?

One solution would be to break apart an image into a discrete grid of bright and dark points. For each point, there could be an electrical signal to represent the brightness value of that point. This solution theoretically works. Part of the system design would be to decide the resolution of your system (i.e. how many points will there be in the grid?) For example, there could be 10 points across (horizontally) and 10 points down (vertically). This gives a grid of 100 (10 x 10) points. Thus, there will be 100 tiny transducers converting light energy to electrical signals, each signal electrically representing the brightness value of its corresponding point.

The monitoring system for this design would have 100 transducers to convert the 100 electrical signals back into light energy at the corresponding brightness values. The main disadvantage to this design is that it would require 100 wires to connect the camera to the monitor, and it would also require that the 100 transducers be extremely well aligned. Both of these facts should be magnified by the fact that a 100 transducer system is extremely poor in resolution. Film resolution, for example, has a resolution of at least 4000 horizontal by 4000 vertical. 4000 x 4000 = 16,000,000 points. The human eye has even higher resolution. Imagine making 16,000,000 connections between a camera and a monitor! It should be noted that the human eye has over {16,000,000} nerve connections between it and the brain.

Processing and transmitting each point separately but at the same time is known as parallel processing because they are all happening simultaneously. It allows ultimate control over every tiny point of light and electrical information but it is an overwhelming prospect to achieve technically.

Another solution

Parallel processing is simple to understand. It is a matter of doing one process to many things at once. All it really requires is brute force. However, you know for a fact that in your current video system, one wire connects your VCR to your monitor, or you camera to your VCR. How could all of the information about all of those points of light be mapped onto ONE ELECTRICAL SIGNAL? It will require more thinking, but the result will be an elegant, one wire system instead of a brute force, thousands-of-wires system.

Nanoseconds and persistence of vision

It is a good thing that human eyes are limited in their ability to detect discrete intervals of time and distance. If eyes had infinite resolution in time and space, we would see literally every electron spinning around every atom at every moment. It would be hard to get anything useful done.

Human persistence of vision is what makes motion pictures work. Like the name suggests, movies are nothing more than a series of still pictures, played one after another, in time. Although there are discrete moments where each picture is displayed, the eye and brain cannot detect when one ends in time and the next begins. The eye is recovering from looking at the previous image when the next one comes on. This fools the eye and brain into thinking there was actually motion between the two images... that there was a whole series of tiny discrete steps between the two images. But there were only two images.

Notice that even though things are happening from nanosecond to nanosecond, humans only see things that happen once every twenty-fourth of a second or so.

Turning light to electricity

The challenge is to map an entire series of brightness values (points) that occur in one moment onto ONE ELECTRICAL SIGNAL. This challenge is overcome by redefining what a "moment" of time means. In the physical universe, a moment of time may mean a picosecond or smaller. But human perception will not detect moments any faster than 1/24th of a second. Therefore, a moment can be redefined as, in the example of motion pictures, 1/24th of a second. This means there are twenty-four moments every second. Each moment is photographed as a separate frame on a strip of film. Then the film is played back, projecting twenty-four frames every second. The human eye cannot tell the difference between 24 moments happening every second and 2,000,000 moments happening every second because of the eyes' persistence of vision.

One transducer... one electrical signal...

One transducer can transduce each point of radiated light from a scene into a corresponding electrical voltage on an electrical signal. Unlike the parallel design in which there was a unique transducer for every point of light, now there is one transducer to do the whole job.

Parallel design does not worry about *when* things happen, because all points of light are transduced in the same moment.

In the serial design of light-to-electricity transduction, each point of light is transduced to electricity a little bit later than the previous point. One point at a time, each point of light is transduced to electricity until an entire framed scene is transduced. Then the process begins again, starting at the first point of light and ending at the last point.

Scanning

The method developed for transducing one point of light at a time in a video camera is known as scanning. It is a serial method because one point follows the next in time. Only one piece of information is transduced and transmitted at a time. This is very different from a parallel system in which many pieces of information are transmitted at once.

The following is one method to achieve scanning of electrical potentials (voltages) across a screen, over time:

An electron gun (a device which emits a consistent stream of electrons) is at the back of the tube. Because the stream is consistent, the voltage of the electron gun is constant. The stream of electrons is aimed directly at the center of the tube.

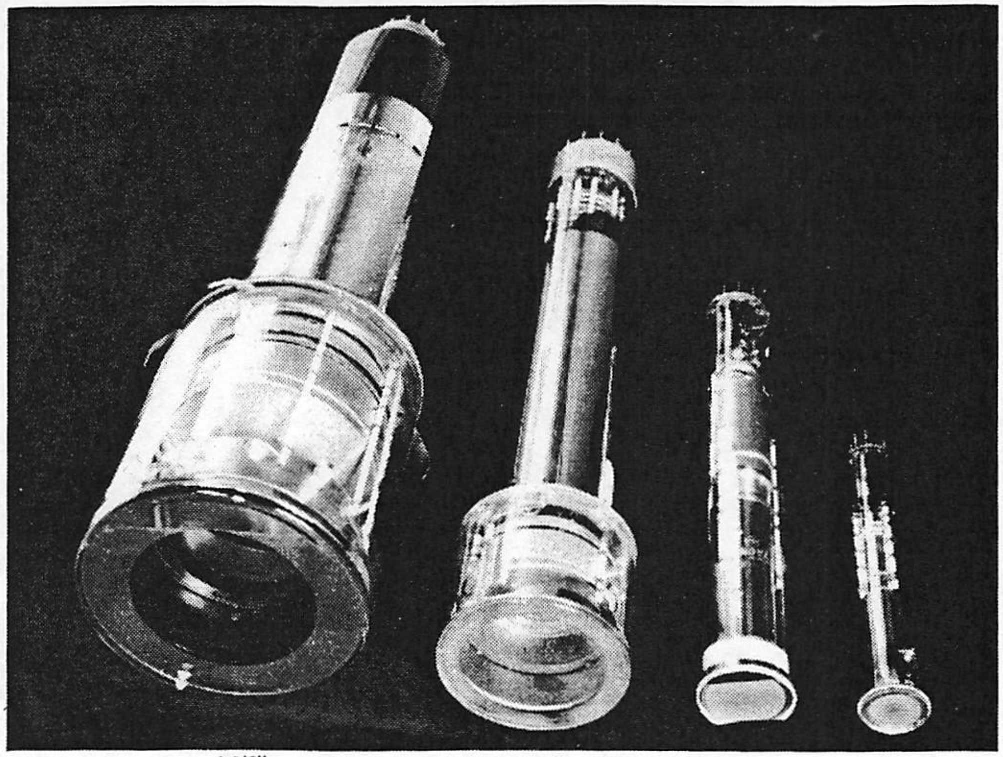


Fig. 1-5. Television-camera pickup tubes.

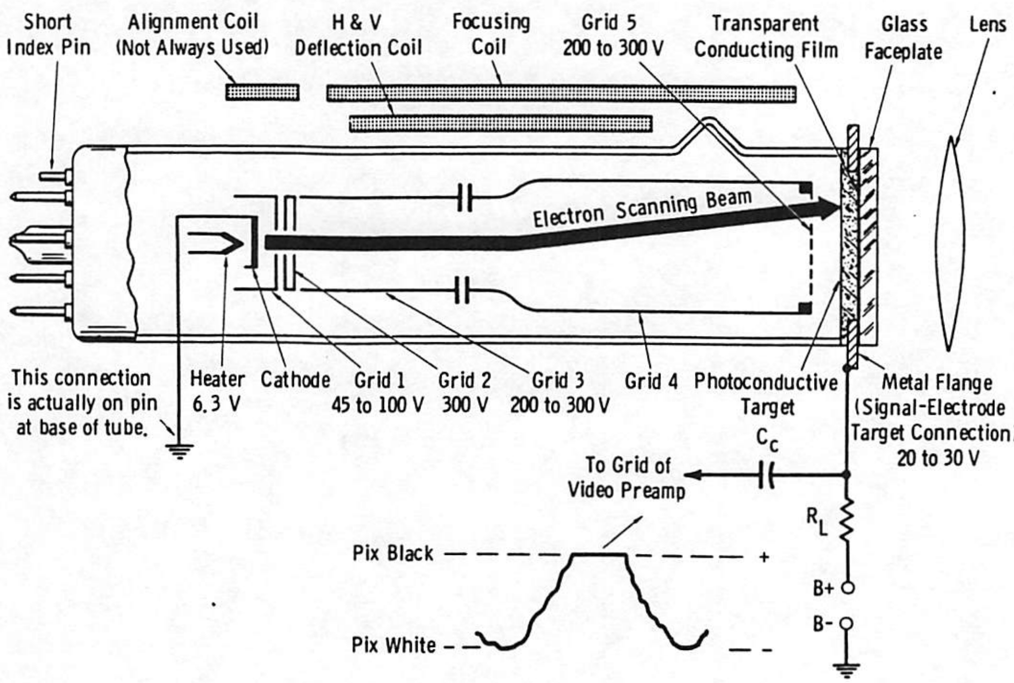


Fig. 1-6. Operation of vidicon tube.

As will be discussed in Chapter 5, in televising motion picture film the image is projected directly onto the target by the lens of the film projector. In this case, the image is right-side up on the target. Direction of scan is easily changed at the sweep coils by reversing the leads from the deflection source, rotation of the yoke, or adjustment of direction of sweep currents. The point to remember is this: The image is said to be scanned from upper left to lower right, which means that *if you were looking directly at the scene in person, you would visualize the scene as being scanned from upper left to lower right.*

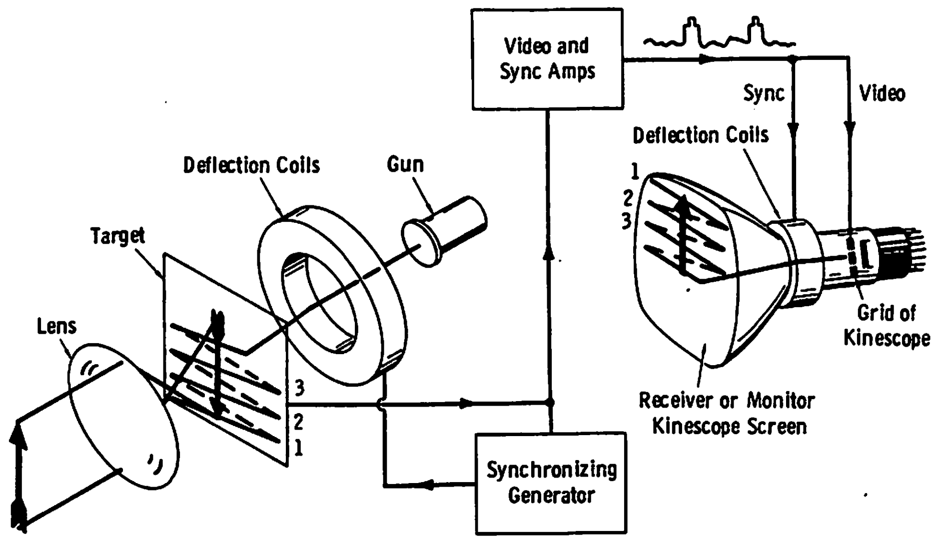


Fig. 1-8. Direction of scan.

The H and V *driving pulses* (synchronizing or timing pulses) are generated in the control room in a part of the main synchronizing generator. They are then fed into individual camera-control units in the video control room; these control units in turn are connected to their respective studio cameras via coaxial cables. Fig. 1-9 illustrates the scanning process. The generator of the sawtooth scanning waveform is generally in the studio-camera pickup head, and is "triggered" in operation by the driving pulses from the studio sync generator.

The question now arises as to why sawtooth waveforms instead of sine waves must be used for the deflecting coils. It should be recalled that a sine wave does not change linearly in amplitude with respect to time; i.e., the slope of the curve at any point depends on the angle corresponding to that point. This would cause the scanning spot (aperture) to move across the image with varying velocity, causing varying brightness across the reproduced picture.

A sawtooth wave is illustrated in Fig. 1-10. Such a wave increases linearly with respect to time (slope of curve constant), returns quickly to the "x" axis, and then repeats the cycle. Such a current waveform through the H and V deflection coils in the yoke about the camera pickup tube

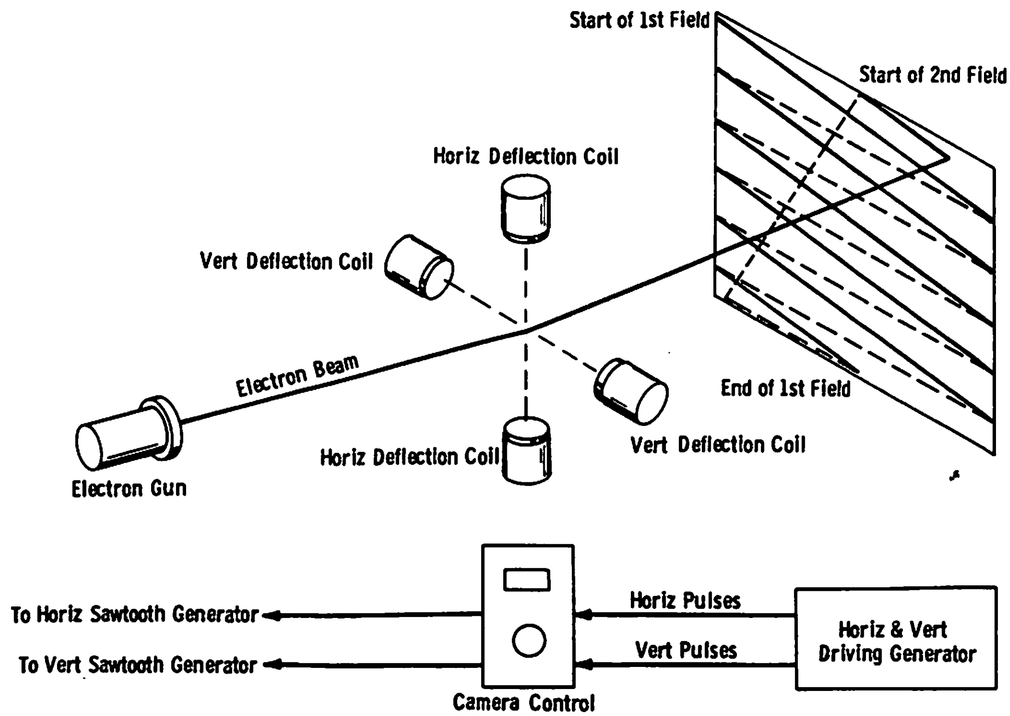
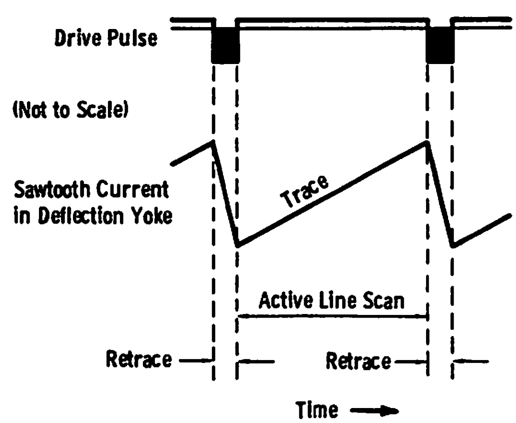


Fig. 1-9. Scanning process.

will cause the scanning spot to move at constant velocity across the scanned surface.

Fig. 1-9 shows the basic principles of the *odd-line, interlaced* scanning system that is standard for modern TV broadcasting. Odd-line scanning means that the total number of scanned lines is an odd number. It may be seen from Fig. 1-9 that this method of scanning allows the spot to return to the top of the surface to start scanning the second field at exactly the same *height* that it had when it started scanning the first field. Fig. 1-11 illustrates the action of the horizontal and vertical sawtooth currents on the scanning process. Current in the horizontal deflection coil causes the electron beam to deflect from left to right, then rapidly retrace as shown. The start of the second line is farther from the top of the raster because the sawtooth current in the vertical coil is lower in value, as shown.

Fig. 1-10. Sawtooth scanning waveform.



To make the beam of electrons scan across the top "line" of the video tube, it is necessary to move the stream of electrons horizontally from left to right. By placing the stream of electrons in a magnetic field, the beam can be moved by making one side of the magnetic field stronger and the other side weaker. By varying which side (left or right) of the magnetic field is stronger, the beam will move HORIZONTALLY according to the strength of the magnetic field. The same technique is used to move the beam VERTICALLY from one "line" to the next. So there are two magnetic fields varying in strength which drive the horizontal movement and the vertical movement of the electron beam.

The purpose of the scanning is to measure the constant voltage of the electron gun against the varying voltages at the surface of the tube. In very quick intervals of time, the electron beam is aimed by magnetic fields at a different point on the surface of the tube. The electrical potential between the electron beam and the point it is currently aimed at on the surface of the tube is the electrical potential (voltage) of the video signal for that moment. In the next moment the gun has moved horizontally to a new point slightly to the right. The voltage between the electron beam and the new point may be different than the previous point because there are more photons (brighter light) hitting that point of the surface of the tube. This means the video signal at this point will have a slightly higher voltage.

It is important to note that there aren't really lines and frames before the electron gun turns on... it is simply a matter of where the electron beam scans that determines lines and frames. The direction and placement of the electron gun is completely dependent on the magnetic fields that control them. Without varying magnetic fields to aim the electron beam, no scanning would take place, so only one point - the center of the video tube - would be transduced. The brightness value of that one point would be transduced into an electrical signal over time, but all of the other points would be ignored.

Scanning is what makes it possible to use on electrical signal to represent all of the points on the surface of a video tube.

One line to the next

After the electron gun has reached the right edge of the "frame" that it is scanning, it must return back to the left edge to scan the next line. This is done by quickly varying the magnetically field horizontally, pushing the electron beam to the start of the next line. The vertical magnetic field also guides the beam slightly lower for the next line to be scanned. Without vertical control, the same line would repeatedly be scanned, like a dot matrix printer with a paper jam, printing over and over on the same line.

During the short time it takes for the electron gun to move to the beginning of the next line, the electron beam is turned off. This means that no electrical signal will come from the transducer at this point, meaning no light information is being represented on the electrical signal during the time the gun is moving to the head of the next line. The time it takes for the gun to move to the beginning of the next line is called the *horizontal blanking interval* because the gun is blanked during this interval of signal.

When the gun reaches the end of the last line it shuts off and the magnetic fields quickly move the gun vertically and horizontally to the beginning of the first line of the frame. The time it takes the gun to move to the top of the next frame is called the *vertical blanking interval* because the gun is blanked during the interval of time when the gun moves to the beginning of the new frame.

A completed scan of the tube is called a *frame*.

Interlacing - fields versus frames
In current television systems, there is method known as interlacing. This method means that in one pass, the electron beam scans only the odd lines: 1, 3, 5, 7... and the next scan transduces only the even lines: 2, 4, 6, 8... The first scan is called Field 1 and it contains the odd lines. The second scan is called Field 2 and it contains the even lines. Each frame is comprised of Field 1 and Field 2.

Because the field rate is double the frame rate, the screen is refreshed more often than if all lines were drawn at once.

<u>Color Video Signals</u>:
•Light is focused by a lens into a prism or filters
•The filters break light into its three primary Additive colors: R, G, B
•A "red" tube only transduces red light into a varying electrical signal
•Likewise with "green" and "blue" tubes
•the result is three separate video signals, Red, Green, and Blue
•the scanning guns must be synchronized or the three color components will not be scanned in time and cannot be reproduced in time. All video signals must be reproducible accurately in the time axis or no picture can be reconstructed! This is why SYNC is crucial and will always be for any audio or video technology!!
•three components are now waiting for further processing

<u>The Color Monitor</u>:
•three color signals, R, G, and B are sent to three electron guns, all synchronized together
•each gun scans and creates varying voltages (electrical potentials) between one end of the cathode ray tube and the photoemmisive surface. When a high voltage is created, many photons are released from the screen surface, which hit your eye, which results in the sensation of brightness.. Likewise, a low or no voltage will cause no photons to be pushed from the screen, the eye will detect none, and darkness will result.

HOW TELEVISION WORKS

Television works by a process called "scanning." At the sending end (the studio, or an outside pickup point) the image to be transmitted is scanned by an exploring element which travels across the image in a succession of descending horizontal lines (see illustration). This scanning process generates an electrical current (signal) which varies in proportion to the light intensity at each point in the image. The signal is sent over a communications channel (wire, cable, or over-the-air) to the receiving point. There a reproducing element generates a spot of light whose intensity varies in proportion to the signal received. Synchronizing signals, which are sent with the picture signal, cause the light spot to travel over the viewing screen in exactly the same pattern as the exploring element travels over the image field. If the spot travels fast enough, the persistence of vision in the human eye will cause the viewer to see a picture on the viewing screen that matches the image at the sending end.

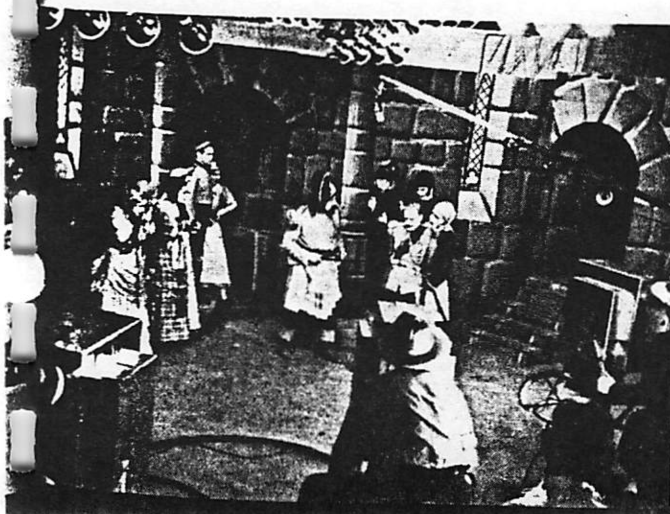
The principle of scanning was suggested by G. R. Carey in 1875, and every television system built to this day has employed scanning in one form or another. The earliest systems were mechanical and used a light-sensitive element for pickup and a light beam for reproduction. Today's systems are all-electronic and use electron beams for scanning, both in the camera pickup tube and in the picture tube. The electronic system provides sharper and brighter pictures. But basically it operates the same way as the early mechanical systems.

For color transmission, three pickup tubes are used (in most systems). Three signals are transmitted (for red, green, and blue images). In an incredibly complex process the receiver separates out the three signals and reproduces them as dots of the three colors. The eye blends these into a single color.

tubes), weighs 45 pounds, and uses only 100 watts of power. Color cameras designed for news gathering are even smaller—the Thompson-CSF Microcam weighs only 8 pounds. In other equipment areas—for example, in studio switchers and in transmitting equipment—there have been equally important (though less visible) advances.

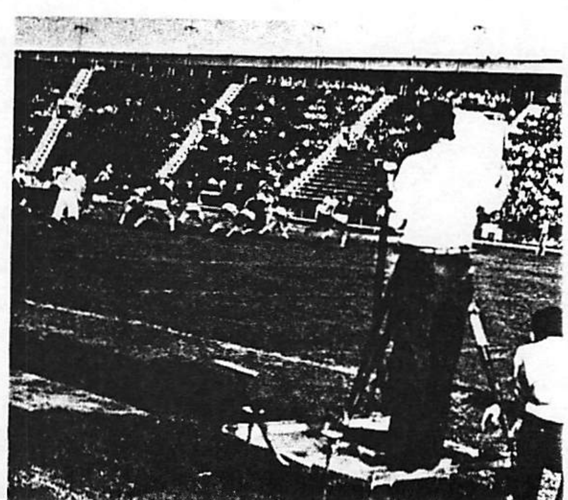
There have been other technological developments which have appreciably affected production and operating procedures. For instance the introduction by Ampex in 1956 of quadruplex videotape recording. Before then all TV programs (except films) were produced live, and usually in the studios of the networks or stations. Today almost all TV programs (except film and news) are taped, and the taping may be done by production companies, in the field, or almost anywhere.

Another development, less spectacular but important, was the introduction—by Philips in 1962—of the Plumbicon®, a new type of camera pickup tube. It



June 20, 1939. TV's first opera, Gilbert and Sullivan's *The Pirates of Penzance*, is the first musical production in NBC's regular TV programming.

September 30, 1939. Fordham defeats Waynesburg State 34–7 in the first televised football game. Baseball, boxing, and tennis will also make their TV debuts this year.



VIDEOTAPE VERSUS FILM

LOUIS JAFFE

All TV that is not live is reproduced from film or videotape. Most television professionals and many home viewers can tell which medium is being used, and most agree that each has a different "feel" as perceived over the tube. Would exactly the same script, performances, camera angles, etc. "look" different on film from the way it looks on tape?

What are the differences between videotape and film as perceived by viewers? Dramatic shows produced on tape are usually described as "flat" or "wooden" compared to similar material on film. Often film is said to be "mellower" and "more human." Conversely, the substitution of videotape for film in news gathering has led to "immediacy."

This "immediacy" of tape seems to enhance documentary TV, but can sometimes work against the sense of fantasy in drama. By contrast, film seems to have a certain "distance" from the reality of the viewer that makes shaky drama more believable.

Scientific study will be necessary to determine whether varying effects of tape and film are due to differences in the way they are received by the human senses. The two do use fundamentally different means to store and retrieve picture and sound. Videotape is an electronic and film a chemical process. Video translates the picture to electronic impulses, which are recorded on magnetic tape; film stores the images directly as a sequence of photographs.

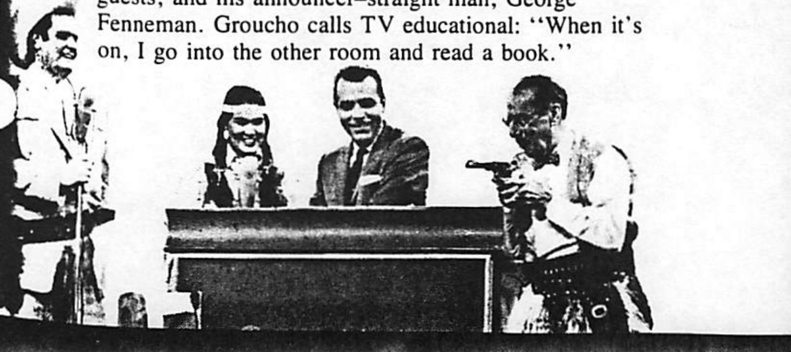
Watching sound film we see twenty-four different pictures a second, interspersed with instants of darkness. In fact, the screen is dark about half the time but the flicker rate meshes with retinal image retention in the human eye and we perceive a persistent picture. This picture is an optical enlargement of the image on the film. Grays in the film image are the light of the projection lamp being blocked by a barrier of silver grains in the film. The light that does get through projects the pattern of the grain which is the fabric of the image. The brightest part of the film image is light passing through clear areas of the film.

In video, the picture is traced by the tip of a moving electron beam, which uses the same scanning pattern the reader's eye uses on a page. Phosphorescence is excited by the passage of the beam in 525 geometrically exact lines. The lines are drawn in two trips from top to bottom of the screen—one for the odd lines, one for the evens. Each 1/60 second trip is called a "field." As one field fades, a second is being drawn. The constantly-regenerating image on the screen is an exact reproduction of how motion is scanned in the camera. The brightest part of this image is the flash set off by the strongest electronic pulse.

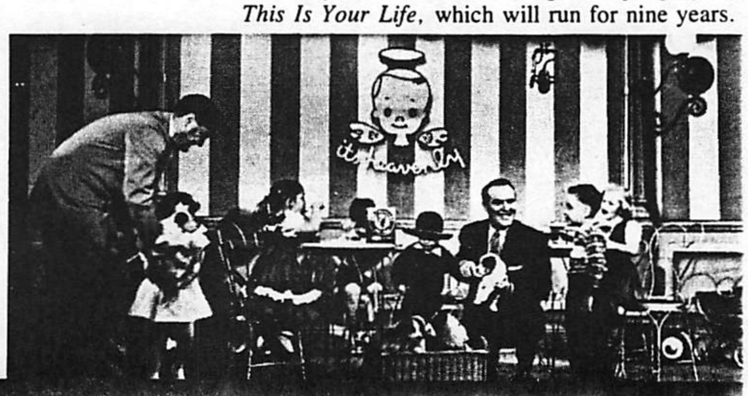
Sound is also reproduced by different means in film and tape. On film, the sound is stored as variations of light and dark which are read by a light-sensing device. Videotape sound is played back magnetically, as in an audio tape recorder. Whether the viewer can hear the difference is, like the question of whether or not he can see the difference between tape and film, a matter of individual sensitivity and professional knowledge.

Louis Jaffe, a New York–based independent producer, works under the name Videotape Projects.

The most unlikely TV game show host is the hilarious Groucho Marx, who begins an eleven-year stint on NBC's *You Bet Your Life*, which he has transferred from radio. The program is enormously popular because of the interplay between Groucho, his strange assortment of guests, and his announcer–straight man, George Fenneman. Groucho calls TV educational: "When it's on, I go into the other room and read a book."



Ralph Edwards is seen on several audience-participation shows in the early fifties. He becomes well known as the first host of the game show *Truth or Consequences* when it appears on NBC in 1950. In two years Edwards will begin his pet project, *This Is Your Life*, which will run for nine years.



Color Video Signals

Early video signals were electrical signals representing brightness information in a scene with horizontal and vertical *sync pulses* every line and field, respectively.

The invention of a color video signal required the analysis of how color is interpreted by the human eye so as to determine a way to encode that information electrically.

The nature of color
Color is determined by the human eye. Different frequencies of the electromagnetic spectrum are interpreted as different colors by the human eye.

Additive color theory
Additive color theory states that any color can be produced by a combination of three primary colors. With Red, Green and Blue as the primary colors of the system, monochrome, white light can be produced in this way:

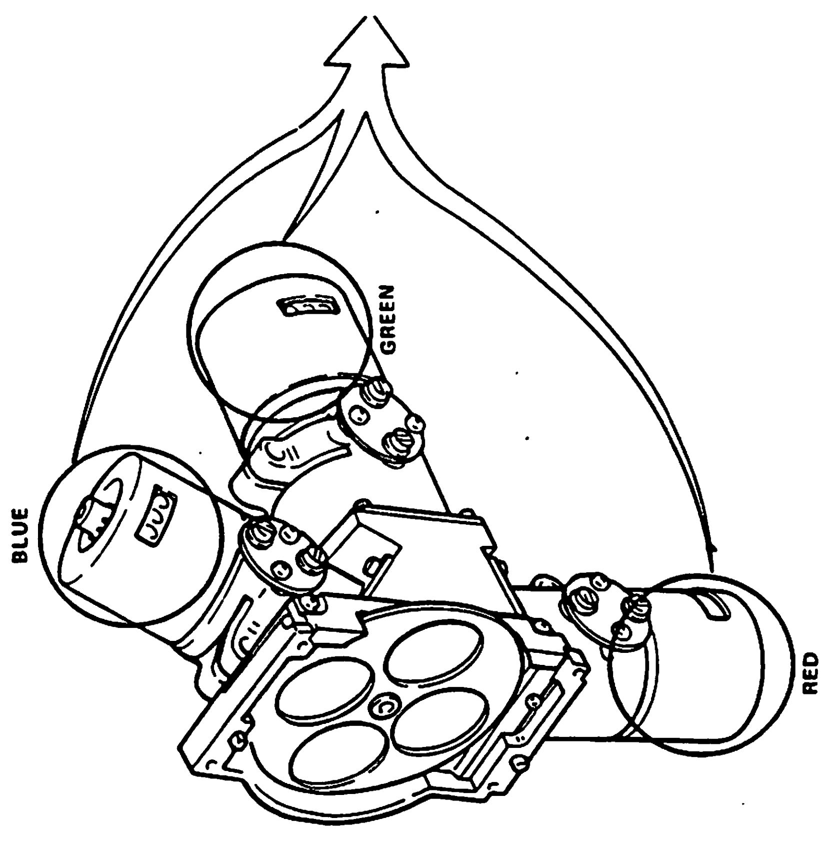
White light = Red, Green and Blue at equal, full strengths
Black (no light) = Red, Green and Blue are all off
Gray light = Red, Green and Blue at equal strengths, somewhere between full and off

A color video camera
A color video camera is a three black-and-white cameras in one.

First, the light from a scene enters the lens of the camera. The light is split equally to a Red filter, a Green filter and a Blue filter. At this point there is light representing only the Red in a scene... light representing only the Green... and light representing only the Blue in a scene.

Three video tubes scan and transduce the Red, Green and Blue light separately. Three color video signals leave the camera instead of one luminance (brightness) signal.

A color video monitor
A color video monitor has three electron guns instead of one, one for each primary color. One color video monitor design has a triple-grouping of a Red, Green, and Blue phosphors for each "point" of light. A mask between the electron guns and the surface of the monitor keep electron beams hitting only their corresponding phosphors. This means that the Green gun, for example, will only activate the Green phosphors as it scans across the screen.


GREEN
BLUE
RED

6 -

Three color components: Red, Green, Blue

A color camera outputs three color *component signals*. A video monitor, likewise, expects to see three color component signals. Connecting a camera's Red output to the monitors Blue input will cause the monitor to be representing Blue information based on the Red light entering the camera. This can be interesting, by certainly not accurate to the original light information in the scene.

The economics of color TV

There is no doubt that a Red, Green and Blue television system works. But it came along after the one luminance channel, black-and-white system was invented. One luminance television signal, complete with sync, was broadcast to televisions across America. The new color TV system obviously required three channels instead of one. This would mean tuning your television set to the Red, Green, and Blue channel just to get one complete color picture. The new system would require everyone to buy new televisions capable of displaying the three color component signals. Economically, this was not feasible.

Now that a black-and-white system was established, how could it be altered to accommodate color information while still retaining the original design of the signal?

Accommodating color to the black-and-white consumers

The new color video signal had to be carried on ONE channel, not three. It also had to have the luminance information AS WELL AS color information.

The first step is to retrieve the luminance signal from the three color components. Recall that in additive color theory, equal values of Red, Green and Blue produce a monochrome signal. An electronic device is necessary to calculate luminance information based on the three primary color components.

This device is called a *matrix* because it performs a mathematical operation on three variables, (R)ed, (G)reen, and (B)lue. The result of the equation within the matrix is a single luminance channel.

The equation of detail, what colors matter most to human eyes?

The most important thing about a black-and-white TV signal is the resolution of the image. How fine is the detail of the image on the screen? The higher the resolution, the better the detail. When converting three color component signals into a black-and-white signal, retaining the highest detail is the goal.

Which colors do human eyes detect the finest level of detail? The science of *colorimetery* studies how human eyes detect color and how well. Colorimetery studies in the first half of the twentieth century determined that humans see the highest level of detail in green light, then red, and finally blue.

The equation for luminance became:

Y (luminance) = 59% Green + 30% Red + 11% Blue.

This is the equation the matrix performs on the three color components to determine luminance. So R, G, and B are encoded into Y (luminance) information.

Once Y is calculated, the signal necessary for black-and-white televisions has been attained. However, if only this signal is transmitted, the color TVs will not be able to determine how much of the original signal was green, red, and blue. It knows 59% of the green signal was used to produce the Y (luminance) signal, for example, but it cannot calculate how much green was in the original scene. More information is needed to determine how much of each color component was in the original scene.

Note that originally, there are three 100% signals: 100% G, 100% R, 100% B). Notice that Y (luminance) is a full strength, 100% signal (59% G +30 % R + 11% B = 100% Y). To be efficient, further math is done to compress the amount of information used to represent the original three signals.

Here is the math done to reduce the amount of information necessary:

Y = 59% G + 30% R + 11% B
R - Y = small value representing the difference between the Red signal and the luminance signal
B - Y = small value representing the difference between the Blue signal and the luminance signal

The result is a luminance channel and two color difference signals: Y, R-Y, B-Y. Once the matrix has performed these operations, a color monitor could do the inverse operations to get back R, G, and B:

R - Y (+ Y) = R, Red signal
B - Y (+ Y) = B, Blue signal
Y (- 30% R - 11% B) = 59% G
Increase Green to 100 % and you now have the three color components R, G, and B derived from Y, R-Y, and B-Y.

Y, R-Y, B-Y is still three color components!
The goal to mix three color components on one video signal is not complete. The next step is to combine R-Y and B-Y together to form one color channel.

R - Y and B - Y produces C (color channel).

Now the video signals have been reduced to two signals: Y and C.

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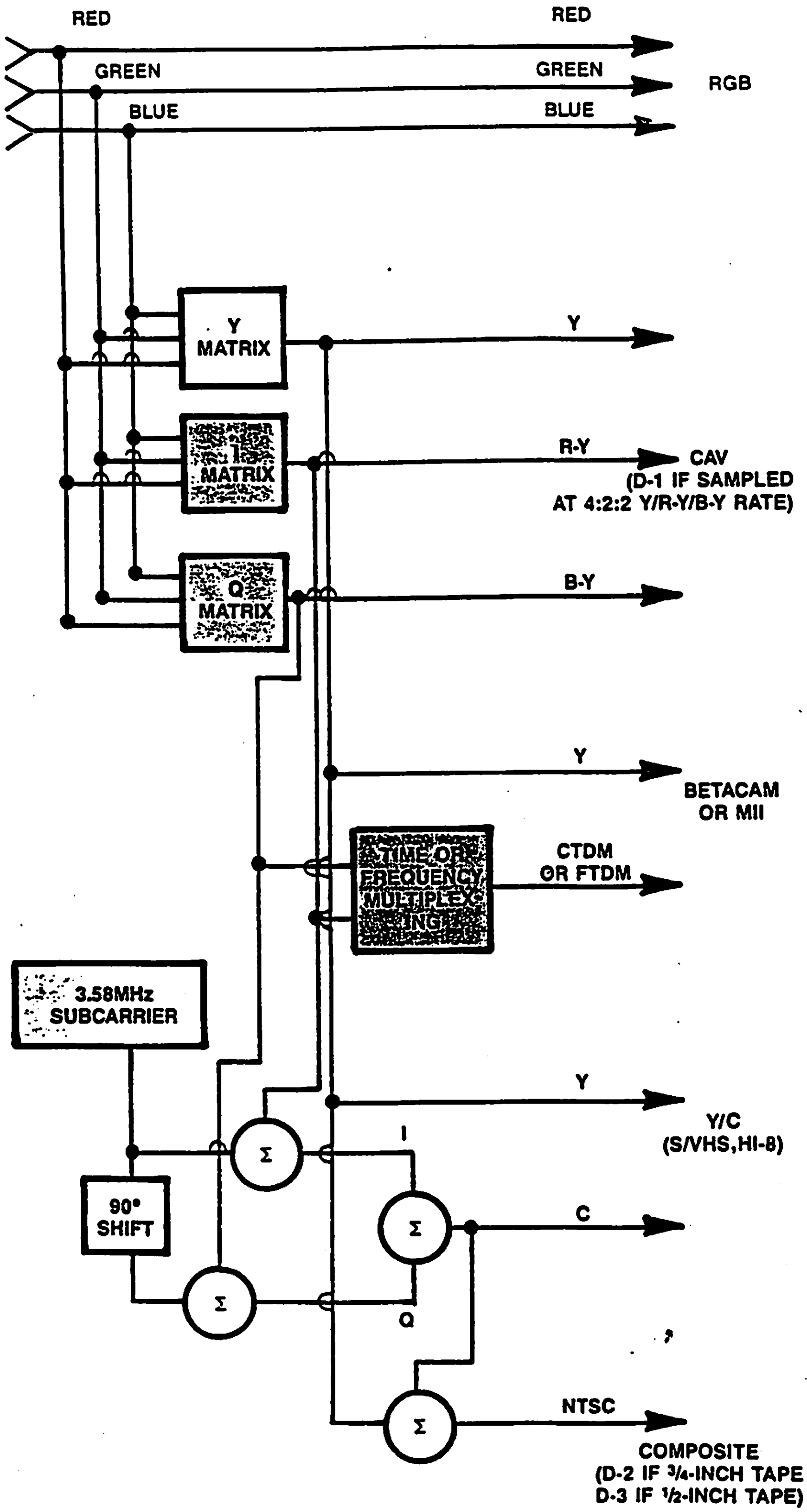
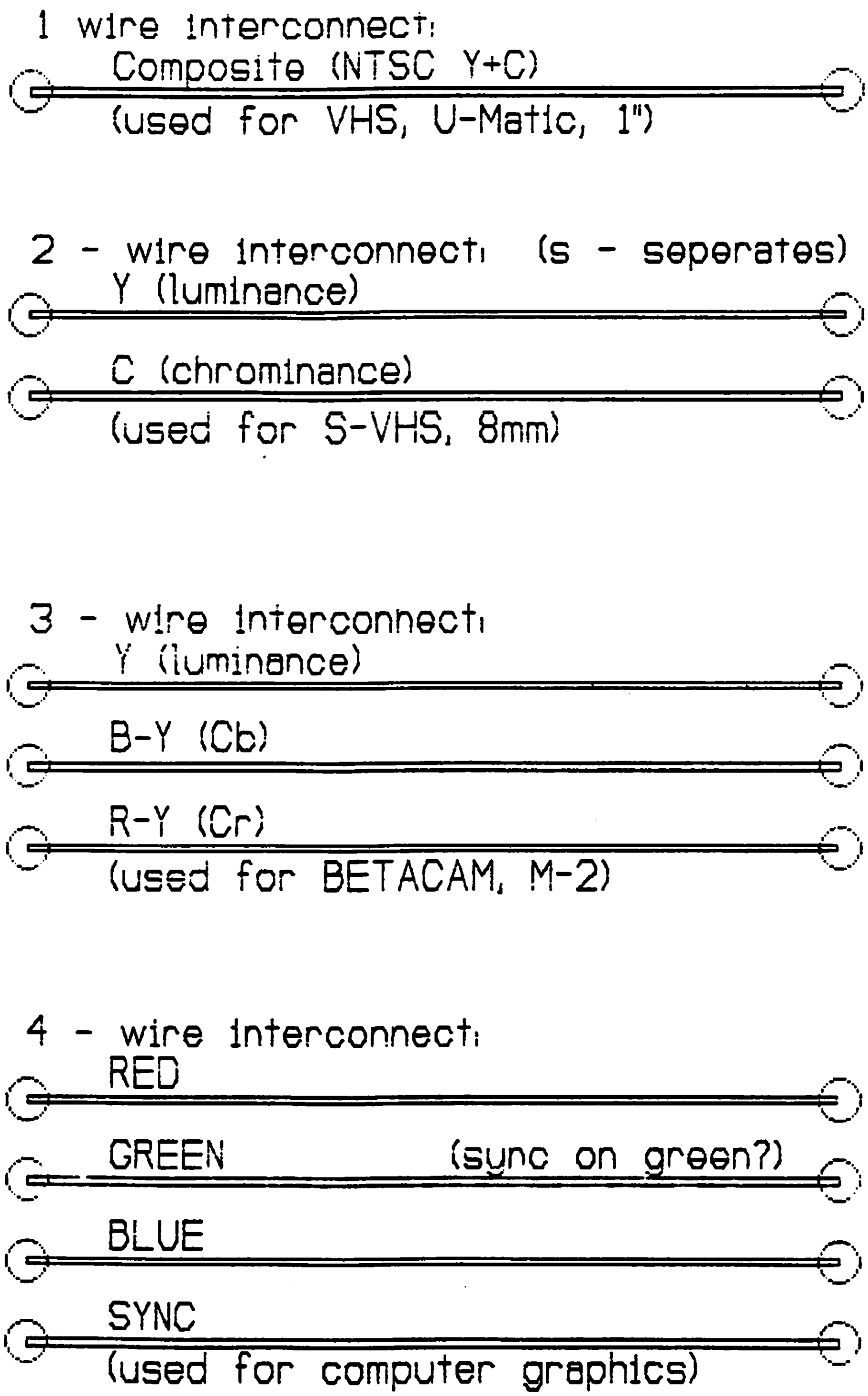


Figure 1. All of today's tape formats can be derived by tapping into the conversion process between RGB and composite formats, such as NTSC and PAL. Today's digital formats are derived by digitizing one of the existing formats.

47


1 wire interconnect:
Composite (NTSC Y+C)
(used for VHS, U-Matic, 1")
2 - wire interconnect: (s - seperates)
Y (luminance)
C (chrominance)
(used for S-VHS, 8mm)
3 - wire interconnect:
Y (luminance)
B-Y (Cb)
R-Y (Cr)
(used for BETACAM, M-2)
4 - wire interconnect:
RED
GREEN
(sync on green?)
BLUE
SYNC
(used for computer graphics)

Still, the processing is not complete. The next step is to combine the Y channel and C channel together. This is done by amplitude modulating the color (chrominance) information on a specific carrier frequency. This carrier is called the *color subcarrier* and it's frequency is 3.58 MHz (3.58 million cycles per second).

Now the luminance channel and the color subcarrier are mixed together, effectively *composited* together. This is why the signal is called *composite video.*

Encoding and decoding color TV

The process of sending R, G, and B signals into the matrix to get Y, R-Y, B-Y and then Y/C and finally composite video is called *encoding* the color information. This happens somewhere from the camera to the broadcast transmission. The signal is broadcast to color TVs and black-and-white simultaneously. Black-and-white TV's do not detect the 3.58 MHz color subcarrier and so only receive luminance information. Color TV's were designed to detect the color subcarrier. For this reason, color TV's have added circuitry to decode the composite video signal back into Y/C, Y, R-Y, B-Y and finally R, G, B which drives the three electron guns.

Video - the Monitor

Now that light information has been successfully coded onto a single electrical signal, you can sit back, relax, and see nothing happen!

Clearly, a video system requires turning the electrical signals back into light so that the eye can view them.

Video monitors - reversing the camera
Video monitors are basically the opposite of cameras. They are glass vacuum tubes (called *cathode ray tubes (CRT)*) with a photoelectric (phosphorescent) coating. There is an electron gun at the back of the tube which scans Field 1 and then Field 2. The electron gun varies its voltage depending on the video signal coming into it, and as it scans it causes proportional numbers of photons to be released from the surface of the screen.

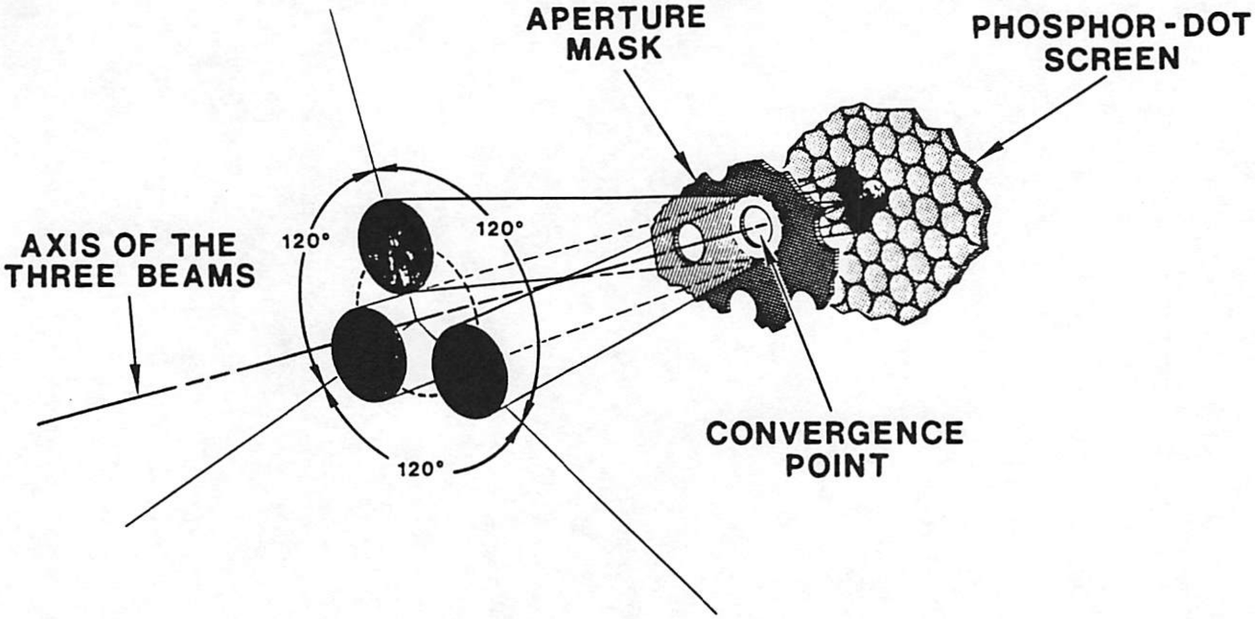
Camera and monitor working together
The camera takes in images by scanning the surface of the video tube and transducing the light energy to electrical energy over time. The monitor takes electrical energy over time and transduces it to light via the phosphorescent coating of the cathode ray tube (monitor screen).

Sending electrical information (which represents the original light information) from the camera to the monitor is not enough! The monitor does not know where in the electrical signal to begin the first line. In other words, the camera is scanning and the monitor is scan - both at the same rate - but they did not necessarily start at the same time. This means the monitor may be scanning the beginning of lines 5 microseconds later than the camera scans the beginning of its lines. Although the monitor is receiving the electrical picture information, it is drawn on the screen five microseconds later than where it was when it started at the camera.

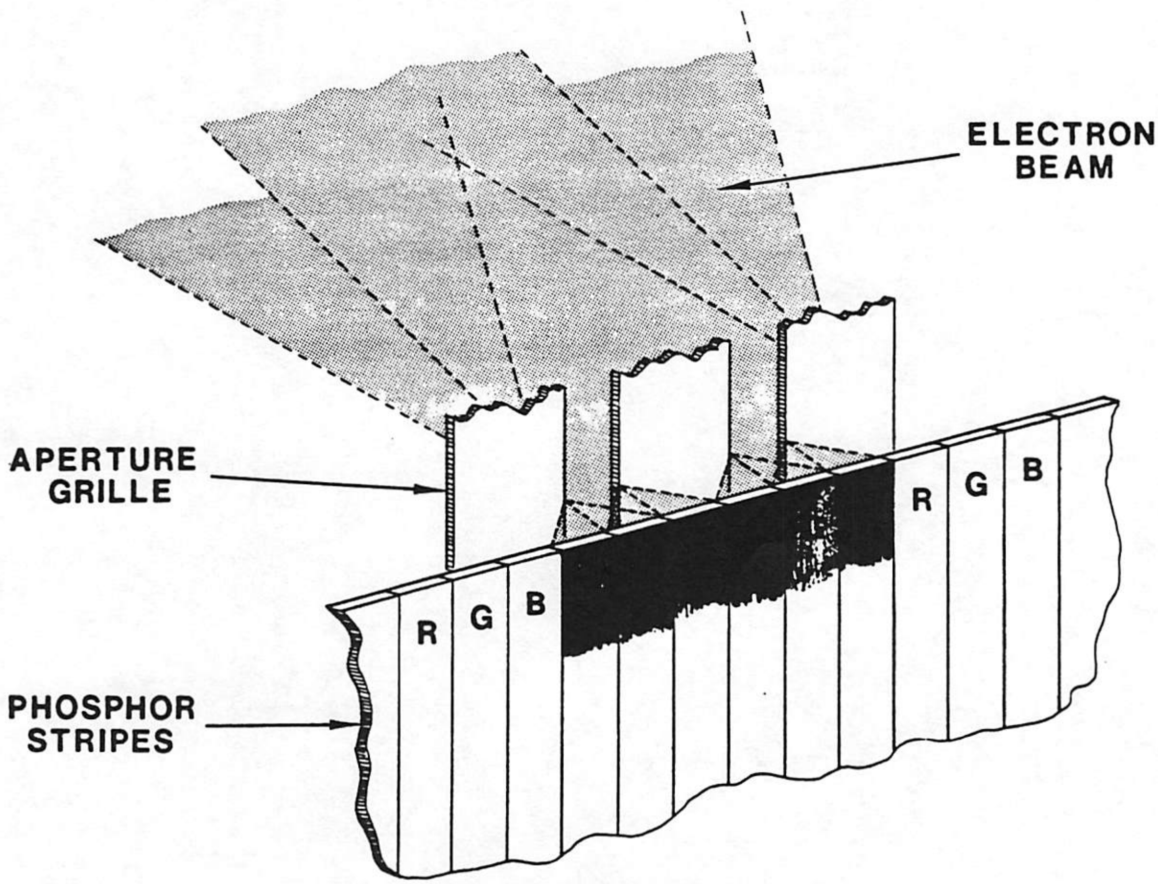
The solution to this problem is *synchronization* . By sending the same pulses to both the camera and monitor designating the start of horizontal scans and vertical scans, the electrical signal produced by the camera will be accurately displayed on the monitor.

Sending synchronization information
It is simple to send the internal horizontal and vertical scan information that a camera uses to drive its electron gun as part of the video signal to the monitor. In this way, a video monitor can synchronize its horizontal and vertical scanning to the camera scanning, and the picture information will then be accurately displayed.

Without synchronization, pictures cannot be put back together on a video monitor. Synchronization is crucial to any video signal! Remember that the picture information still exists in the video signal, but there is no way of coherently displaying it without the sync information.



SHADOW MASK



TRINITRON

A VISIT WITH VLADIMIR ZWORYKIN, THE FATHER OF TELEVISION

JOHN P. TAYLOR

How does it feel to be a legend in one's own time? Dr. Zworykin simply shrugged off the question. He doesn't think of himself that way. He is aware, of course, that many refer to him as the father of television—the man who made it possible—the inventer of the iconoscope and the kinescope, and so on. But he pays little heed. And, unless prompted, he seldom refers to the past. His consuming interest is the present—and, even more, the future. It is symptomatic of this remarkable man.

But Vladimir Kosma Zworykin, whether he recognizes it or not, is a legendary figure in electronics. Born in Murom, Russia, in 1889, his life has paralleled—and been closely entwined with—the long evolution of television. His interest in electronic television began in 1907, when he became a laboratory assistant to the famous Professor Boris Rosing. He graduated from the Technological Institute of St. Petersburg in 1912, served as a communications officer in the Czarist army during World War I, came to the United States in 1919. In 1923, he patented the iconoscope—the breakthrough that made electronic television possible. In the thirties he played a major role in RCA's long struggle to bring television to the public. During World War II his group worked on infrared devices (the snooperscope, etc.) and on television-guided missiles. In 1954—having reached the mandatory age—he officially retired from RCA and immediately began a new career with the Medical Electronics Center of the Rockefeller Institute.

> "How does it feel to be a legend in one's own time?"

Today, at age 88, one might expect to find him rocking on his porch and musing on the past. Not Dr. Zworykin! We found him in his office at RCA laboratories in Princeton, where he spends almost every morning. Afternoons he is apt to be at the Fusion Energy Corporation, also in Princeton, where he is active as a consultant. He is also a consultant on medical electronics to several organizations. And, from December to April, he is a "visiting scientist" at the University of Miami.

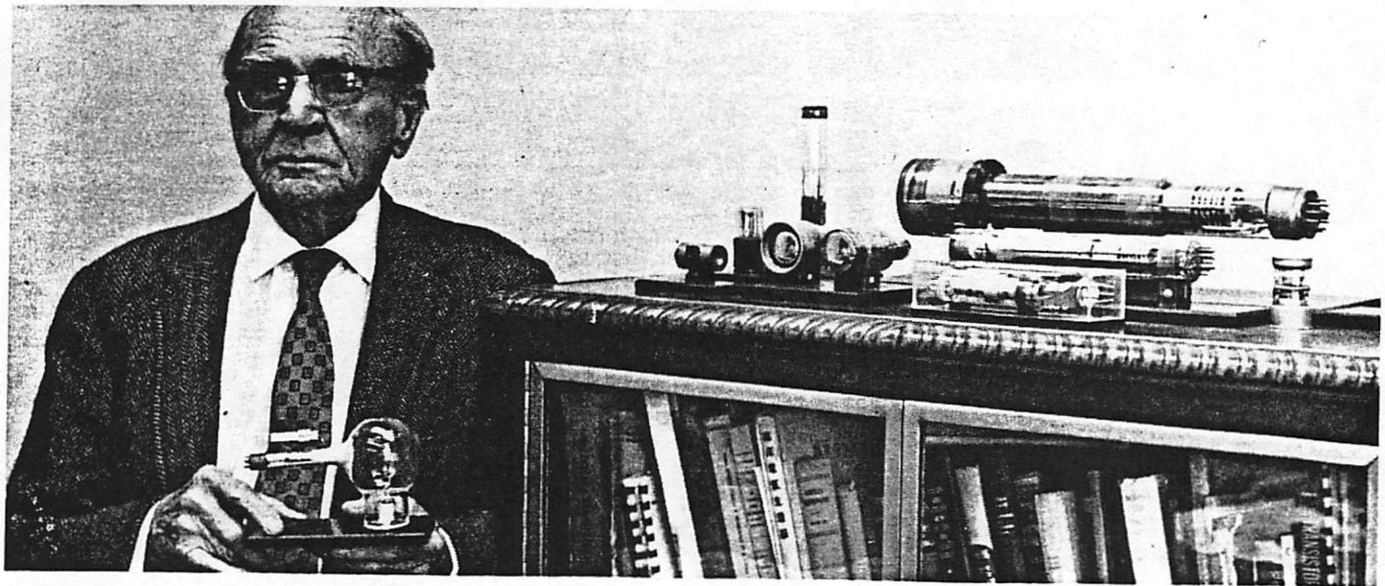
Dr. Zworykin does not consider all this activity unusual. He says, "When you work all your life you can't stop. What can you do, play golf? It's too late for me. Drinking I never did, just a little stuff. I don't smoke. You have to occupy yourself." And he does!

His office at RCA (where he is an honorary vice president) is very much a working office. None of his fifty or so awards adorn the walls. There is only one picture—a small photo of B. J. Thompson, a brilliant associate who was killed while testing radar over occupied Italy. But there are papers and reports and books in great profusion. Most are in the area of theoretical physics, which has been Dr. Zworykin's

In 1947, Jack Barry begins a seven-year stint as host of the entertaining children's program, *Juvenile Jury*. In the fifties, he will also moderate another kiddie show, *Winky Dink and You*.

December 27, 1947. "What time is it, boys and girls?" *Howdy Doody* begins a thirteen-year run on NBC as a daily fifteen-minute program.

Puppet shows abound from coast to coast. *Kukla, Fran, and Ollie* debuts locally on WBKB in Chicago in 1947. Burr Tillstrom is the puppeteer for the Kuklapolitans who visit Fran Allison.



Dr. Vladimir Zworykin poses in his office at RCA in Princeton, New Jersey with his 1923 invention, the iconoscope.

love since he was a very young man, and which he still follows avidly.

Fortunately for us, Dr. Zworykin, despite his preoccupied air and his imposing presence, is a warm and kindly man. He greeted us in the friendliest fashion and patiently tried to answer our mundane (for him) questions.

How does Dr. Zworykin feel about what has happened to the miracle he wrought? "I feel elated," he says, "especially at times when I see things like the moon landings and, even more, the pictures which the Viking lander sent back from Mars. Without television this would not have been possible."

Has television worked out the way he thought it would back in 1923? "Not exactly," he says. "In those days when they asked me, 'what are you going to do with it?' I said, with television you can supplement your eyes. Television will help the eye where the eye cannot give us the information—too far, too dangerous, too small, and so forth."

The closeups of the moon are a good example of what Dr. Zworykin was thinking of fifty years ago. So too is the use of television to observe nuclear operations, to explore the ocean bottom, to see particles of molecular size (through the electron scanning microscope). Today there are many applications of television in industry, medicine, and education that fit his original concept of "extending man's sight."

Apparently entertainment programs were not an important part of Zworykin's early thinking on television. Even today he is equivocal about them. "I don't use television myself," he says. "My wife looks at it and calls my attention to programs I like—news, wildlife programs, things like that. But when they put on an old story about Amos and Andy as a whole program. . . !"

What does Dr. Zworykin foresee in the future for television? "I think the Moon and Mars telecasts

John Cameron Swayze becomes TV's first big newscaster in 1947 on NBC's *The Camel News Camera*. In 1948, Swayze will also host a panel show called *Who Said That?* and in the fifties he will even do a little acting. Ironically, Swayze will gain TV immortality by losing a Timex watch he is esting for durability in a large tank of water on a ive commercial.

Douglas Edwards is shifted from CBS-radio to CBS-TV to do daily fifteen-minute broadcasts. Edwards will remain the top CBS newscaster until 1962. Because of their lack of popularity with viewers, there will be fewer news programs in 1952 than in 1948.

Gene Autry, the movies' first singing cowboy in the 1930's, leads his horse, Champion, onto the TV screen in 1947. Already wealthy from his numerous Saturday morning matinees and recordings of such songs as "Silver Haired Daddy of Mine" and "Rudolph the Red-Nosed Reindeer," Autry will make it big in television as well.

"I said with television you can supplement your eyes . . ."

open—even for commercial people—tremendous new fields for television. I visualize an international educational and technological library using television and satellites—and perhaps video disks, which can be produced at very low cost, almost the same as music." He thinks satellites will be much less expensive when they are launched from the space shuttle; that possibly they will be battery-operated, with the battery being recharged by the shuttle. At some time in the not-too-distant future he believes all TV will be by satellite.

What about the use of fiber optics for television transmission? "But why?" says Dr. Zworykin. "That won't change the situation. Maybe in the next ten years they will put in cable with fiber optics. But that is just improving the technique which already exists, for specific reasons—cheaper, less cable, etc. It's a good development, but it is not fundamental. What I'm trying to do is something that is fundamental."

What is Zworykin working on? "Nothing. I'm retired," he says, eyes twinkling. "If you ask what am I doing—reading, reading till I'm going half crazy." And, one might add—thinking, thinking, and spouting new ideas faster than anyone can keep up with him.

One of the things Dr. Zworykin has been thinking about is the energy crisis. He believes fusion power is the right long-term answer (*fusion* is the combining of atomic nuclei, as contrasted to *fission*, the splitting of atoms, which is the process used in present nuclear plants). But he believes present efforts to develop fusion power are wrongly directed.

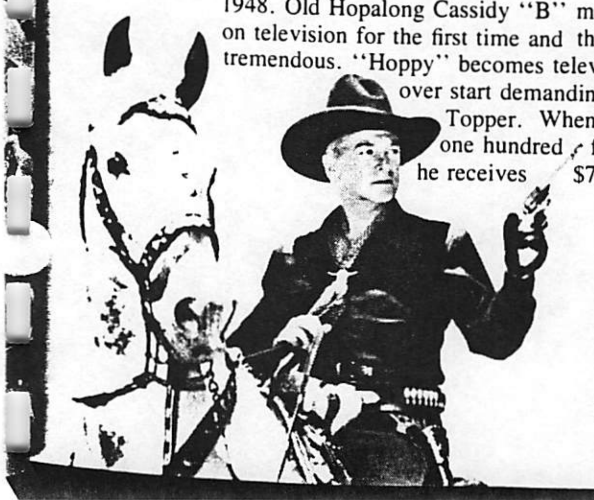
As Dr. Zworykin explains it, "At present they are going to bigger and bigger stations and getting bigger and bigger difficulties. I think that is the wrong way. My idea is to build the generator on a very small scale—something like the laundromat or the electric heater in the house. Make it small, like we do in the laboratory. Why is this possible? Because the energy of particles is tremendous. Some of the work we are doing in television, like the electron multiplier, can be used for ion multiplication on a small scale—like a television tube. Even if you release only one kilowatt, it will be a small device and you can multiply the power by placing several in parallel. Recent figures show that the average family consumption of power is under eight kilowatts, so if you produce up to ten kilowatts then we are independent of the big power stations. Imagine such a device in every family kitchen—all our social order will change.

"It's a dream all right," Zworykin concedes, "but a dream which is based on what is going on right now, and where it is going to."

Actually, fusion has been accomplished in the laboratory on a very small scale. The process is still inefficient—more power in than out. But the possibility is there. And, as Dr. Zworykin points out, if the goal were a small home-type generator (instead of a mammoth power station), probably many small experimenters would be working on it. They would face many problems, but no more than electronic television seemed to face before Zworykin's breakthrough invention of the iconoscope in 1923.

A nuclear generator in every kitchen is only one of Dr. Zworykin's current interests. There are many others—all far-out. For, as Robert Sarnoff said of him at his eightieth birthday party, "His brilliant mind has never waited for others."

John P. Taylor is a free-lance writer and contributing editor to Television/Radio Age.



1948. Old Hopalong Cassidy "B" movies starring Bill Boyd appear on television for the first time and their impact on the youngsters is tremendous. "Hoppy" becomes television's first hero, and kids all over start demanding Cassidy outfits and horses like Topper. When Boyd sells TV rights to his one hundred feature films several years later, he receives $70 million.

Cowboy movies become a major attraction for children, who are introduced to old-time cowboys such as Roy Rogers (seen here with Gabby Hayes), Buck Jones, Lash LaRue, Hoot Gibson, Tex Ritter, and Tim McCoy. By the time Rogers has his own TV show in the early fifties he is already an idol with the new generation.

Dubbing

Once the art of dubbing is mastered, a large portion of the video world can be comprehended. All of the intricacies of specific decks must be understood, track layouts on different video formats, standard specifications of video signals, and how to physically connect decks together. Remember that all of the connections you make are actually electrical connections. All video is electrical by design.

Learning specific video deck features

The best way to learn a video deck is to sit with it for about an hour and identify what all of the knobs and switches and what they do. Remember that each switch or knob was installed to perform some task that the manufacturer, at least, thought was important.

Also look at the back of a video deck and identify all of the connectors - inputs and outputs. The typical inputs and outputs on a video deck are:

Signal	Connector	Purpose of Connection
Component video: Y, R-Y, B-Y	BNC (x3)	BetacamSP video
Y/C (S-Video)	Y/C (S-Video)	Hi-8, SVHS
Composite video	BNC, RCA	VHS, 3/4", 1", video monitor outputs, video or sync reference signal
CTDM (Component time-division multiplexing)	CTDM	BetacamSP DUB video
Dub (Y/C-688)	Dub	3/4" U-matic
Serial Digital video (component)	BNC	Component serial digital: Digital Betacam, D-1, Quantel, Abekas switcher, DVE's
Serial Digital video (composite)	BNC	Composite serial digital: D-2
Audio	XLR	Professional audio level
Audio	RCA	Consumer audio level
Digital audio (AES/EBU)	XLR	2 channels per connector, professional standard
Digital audio (S/PDIF)	RCA	2 channels per connector, consumer standard
Audio monitor	Mini, RCA	Mixed audio output for monitoring
Time code	XLR, BNC, RCA	Time code signals as audio signal
Remote connection	9-PIN, 33-PIN, 45-PIN	Remote control protocol (RS-422 or RS-232) to connect to edit controller or another VTR
AC power	AC cord	AC power to VTR's power supply

Track layouts
Every video tape format has a different way of recording track information onto the magnetic tape, but there are always specific types of tracks:

Video tracks: these are typically written diagonally across the video tape and represent the picture information.

Audio tracks: anywhere from 1 to 4 tracks. Two are laid longitudinally and additional ones are laid in the same portion as the video track, but at a higher frequency so as not to interfere with the video information.

Time code tracks: a longitudinal audio track that records time code as audio information

Control track: all video tapes require a control track (an series of pulses indicating the speed at which the VTR motor moved during the recording process). The control track assures that the VTR will play back the video signal at the same speed it was recorded.

Transcoding signals
Because different video formats record the video signal differently, dubbing from one format to another may require translating one coded video signal to a different coded video signal. This process is known as transcoding.

For example, BetacamSP has Y, R-Y, B-Y outputs and Hi-8 has a Y/C (S-video) input. These two signals are not readily compatible, but both are higher quality than the lowest common denominator composite NTSC signal. All video decks can be connected via composite NTSC, but this sacrifices discrete, higher quality component signals.

To translate between Y, R-Y, B-Y and Y/C, a device called a *transcoder* is necessary. A transcoder has a number of possible signal format inputs and a number of outputs. You can then select between the input format and output format as needed. This is a crucial device when doing interformat dubbing.

Matching audio signals
The high-end XLR audio connectors use three wires: hot, neutral and ground. The hot carries the original signal and the neutral carries the same signal 180 degrees out of phase. Any noise that gets on the wires can be isolated by taking the difference between the hot and the neutral signals. The result will be zero plus any noise that was on the line. The third connector is for an electrical ground connection.

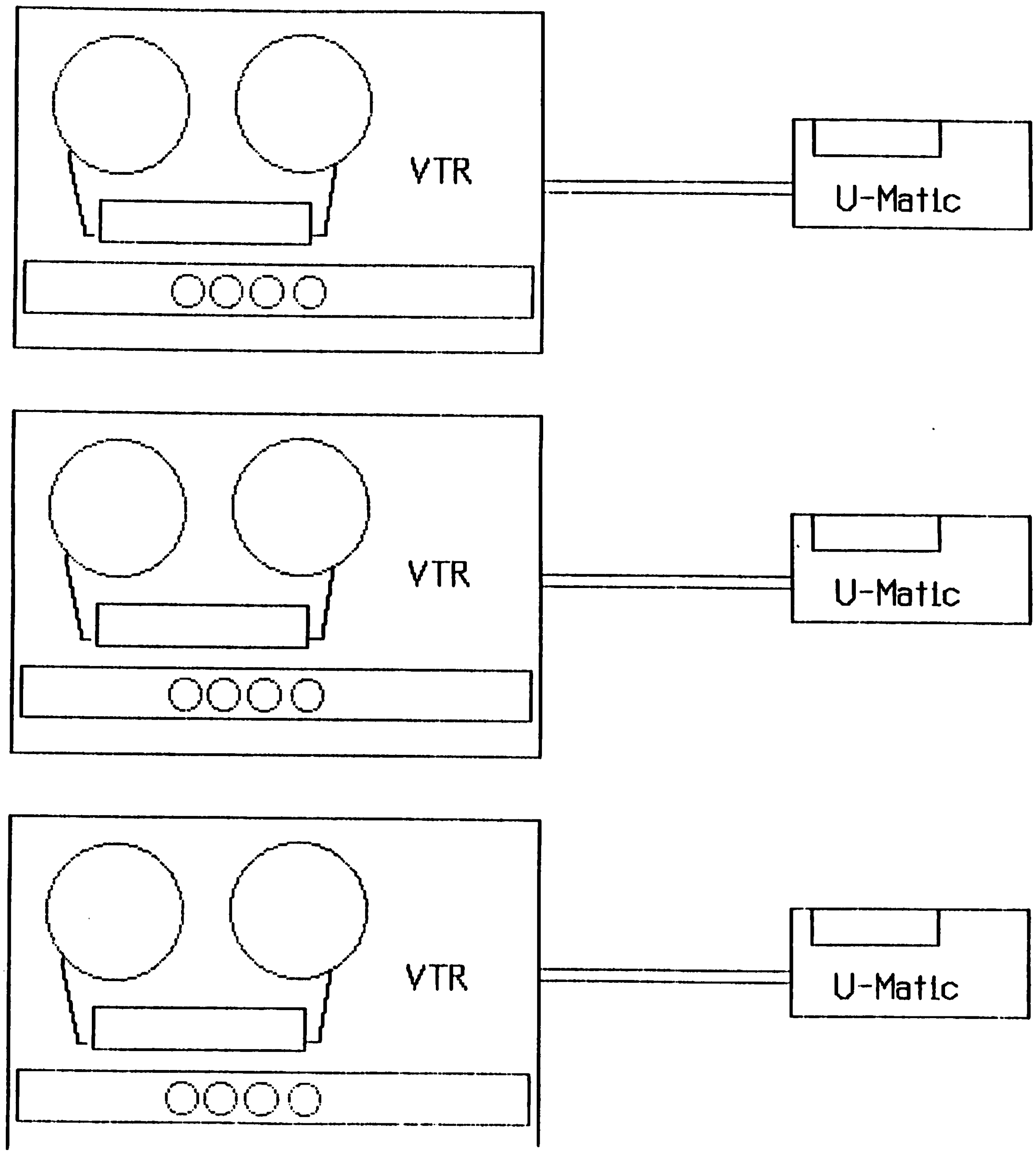
Consumer decks (like VHS) have two pin (ground and signal) RCA connectors.

So, to connect VHS decks to professional decks, adapters like XLR-to-RCA are needed.

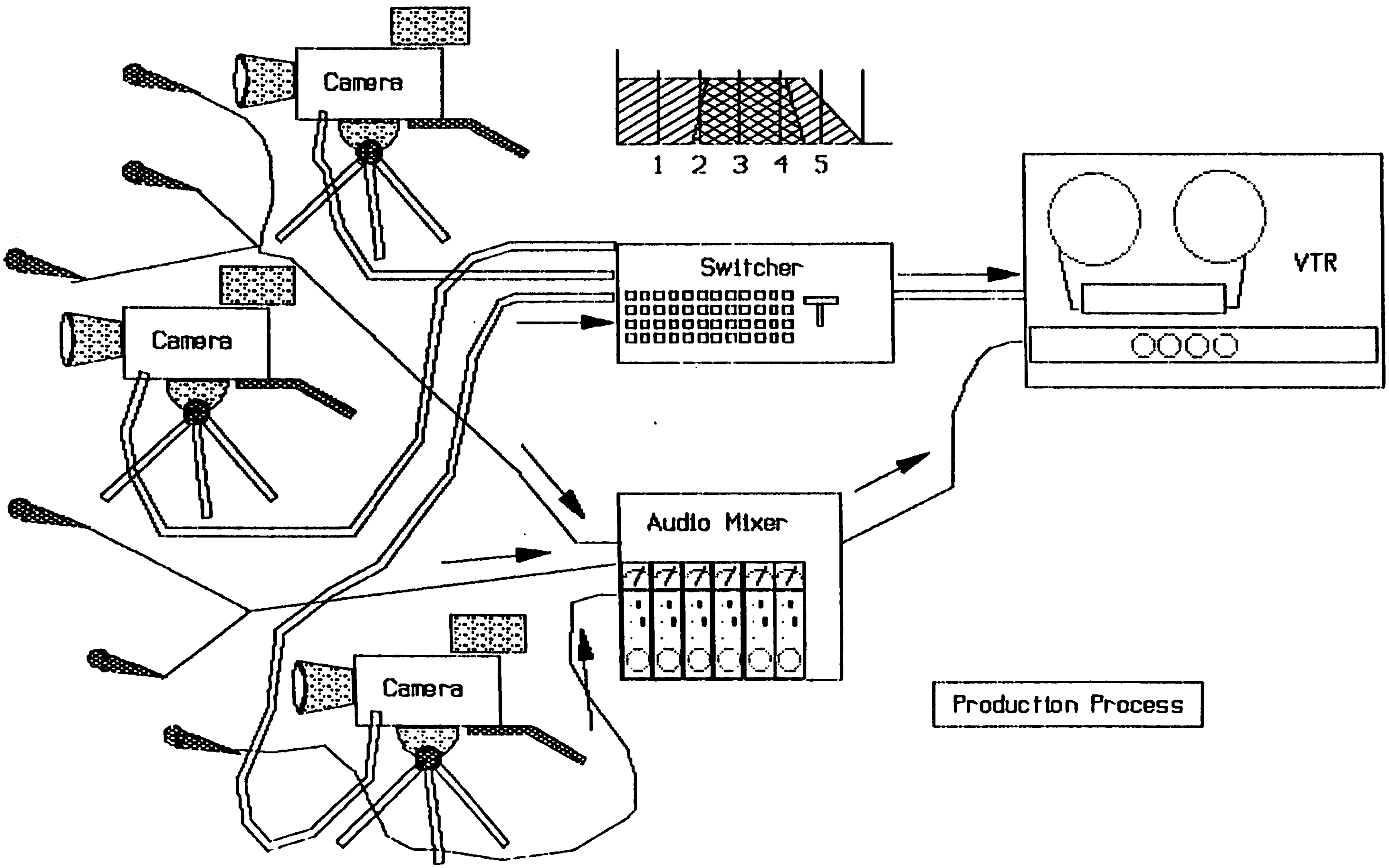
Furthermore, professional decks input and output audio signals at a different level than consumer decks. They are distinguished by the amount of impedance they have against an alternating signal (the electrical audio signal). Therefore, an impedance match must be made.

Connecting consumer audio to professional means matching the impedance, adapting the connecting, and going from hot-neutral-ground ('balanced") to signal-ground ("unbalanced").

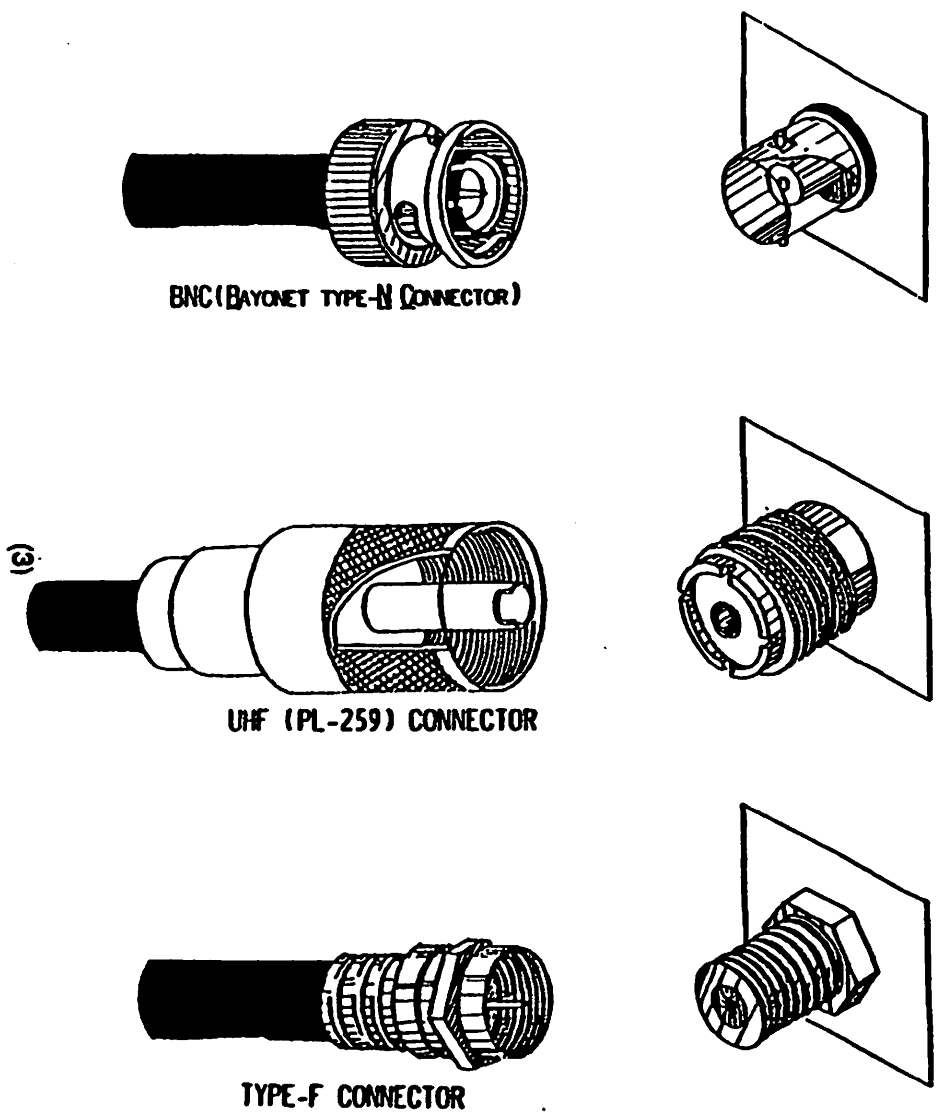
Devices called *matchboxes* meet all three of the necessary requirements.

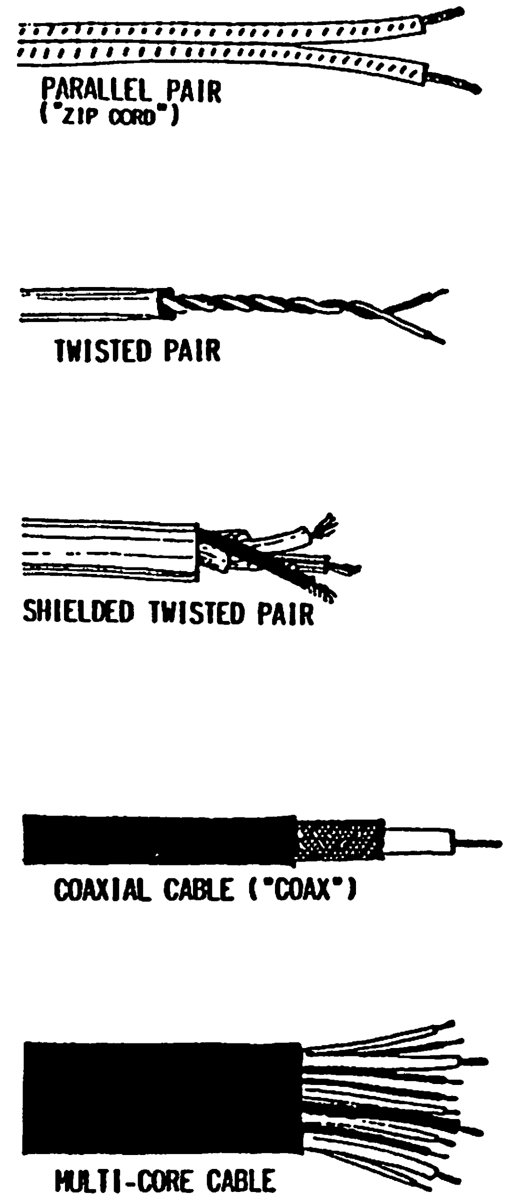


Dub down.

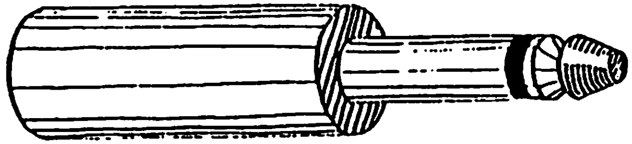

Camera
1 2 3 4 5
Switcher
VTR
Camera
Audio Mixer
Camera
Production Process

COAXIAL CONNECTORS

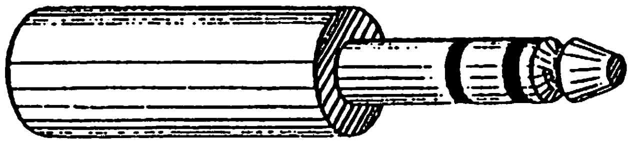




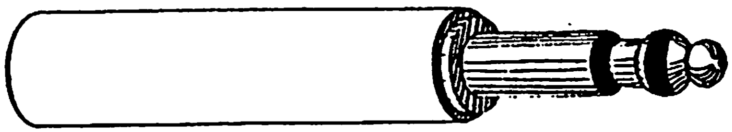
MALE MONO 1/4" (PHONE) CONNECTOR



MALE STEREO 1/4" (PHONE) CONNECTOR



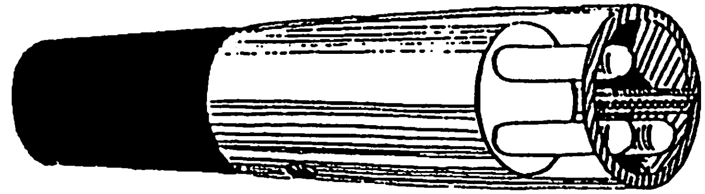
MALE NON-SHORTING STEREO 1/4" CONNECTOR



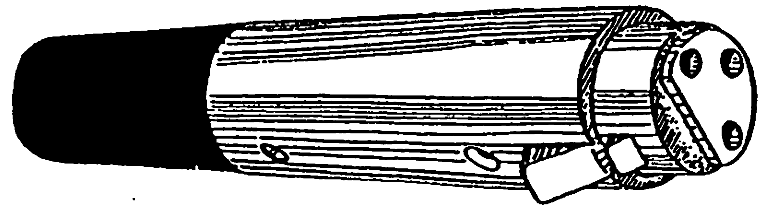
MALE MINI-PLUG CONNECTOR

MALE SUB-MINI CONNECTOR

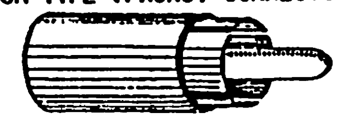
MALE XLR (CANNON) CONNECTOR

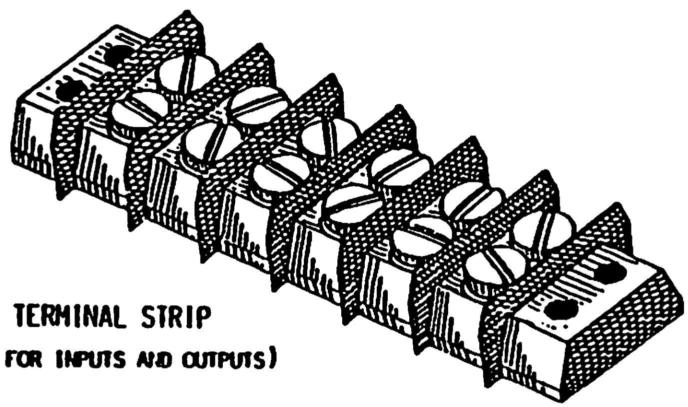


FEMALE XLR (CANNON) CONNECTOR



MALE RCA-TYPE (PHONO) CONNECTOR

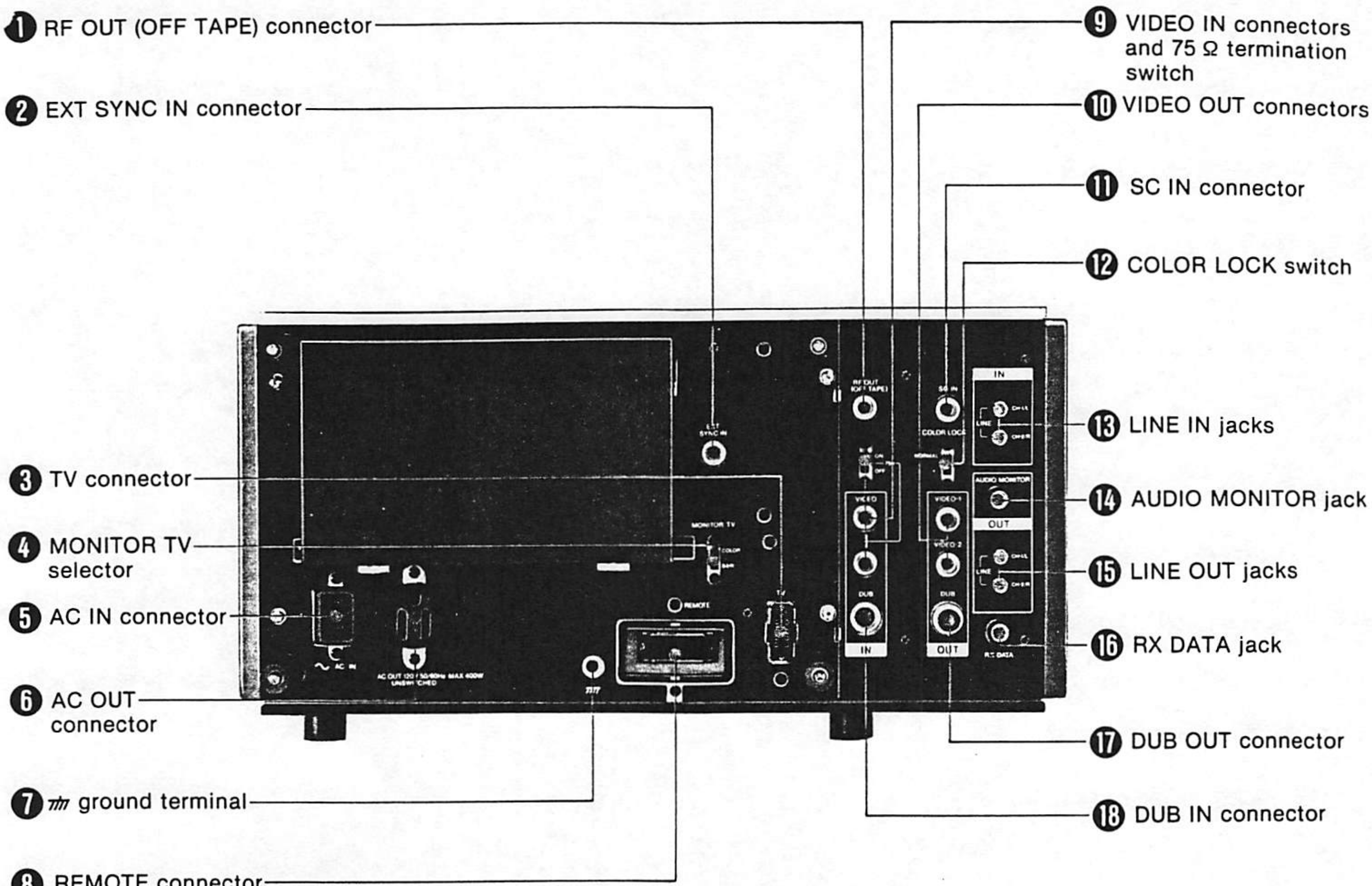




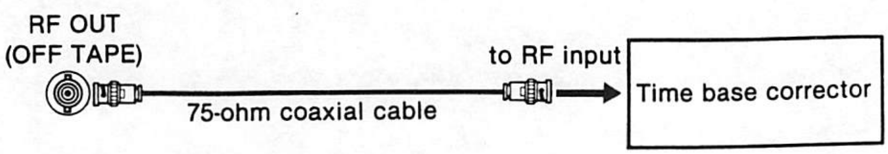
TERMINAL STRIP
(USED FOR INPUTS AND OUTPUTS)

C7

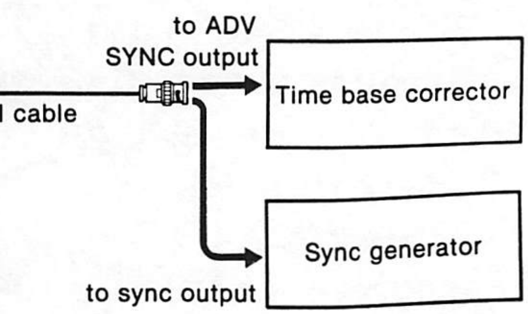
REAR PANEL

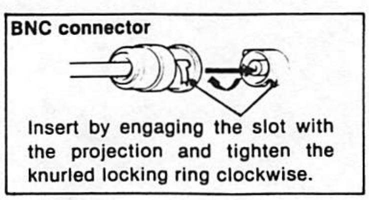


❶ **RF OUT (OFF TAPE) connector** (BNC type)
This connector supplies the FM output signal to a time base corrector.

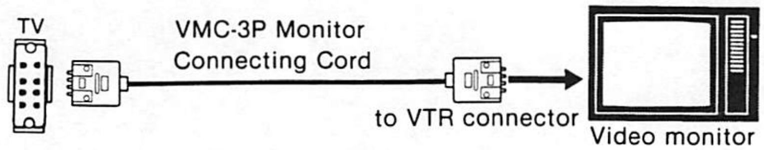


❷ **EXT SYNC IN connector** (BNC type)
Connect to the advanced sync signal output of a time base corrector or an external sync generator. The recorder will synchronize with this signal.

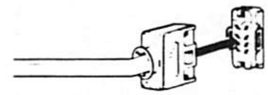




❸ TV connector (8-pin)
Connect to the 8-pin VTR connector of a video monitor.
The video and audio input and output connections can be made with a single cable. When this connector is used, the audio signal will be recorded on audio channel 2. During playback the channel selected by the AUDIO MONITOR switch will be heard through the speaker on the video monitor.



Be sure that the connectors are properly aligned before connecting the cable.



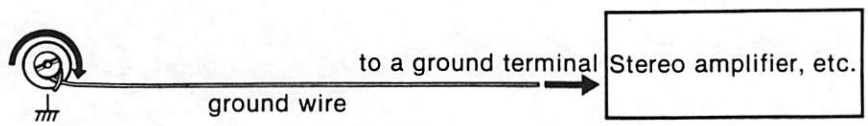
To remove, press the buttons on both sides of the connector.

❹ MONITOR TV selector
Set this selector according to the type of video monitor used.
COLOR: For a color monitor.
B & W: For a black and white monitor.

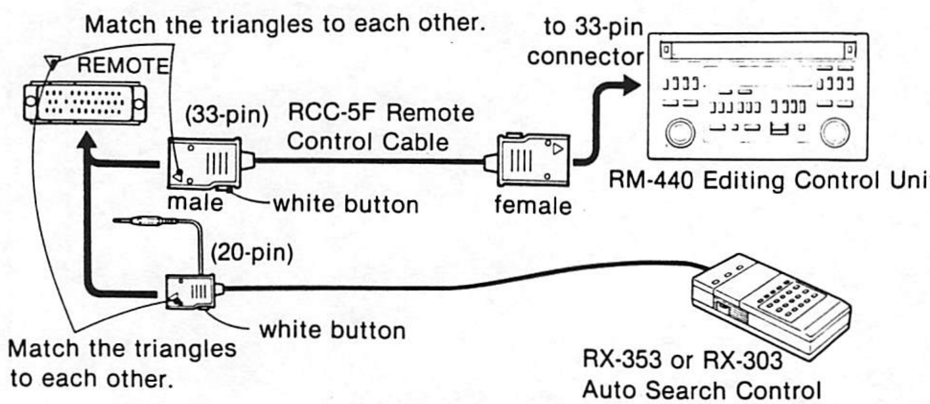
❺ AC IN connector
Connect the supplied ac power cord here.

❻ AC OUT connector
This connector supplies ac power of up to 400 W to other video equipment. (The POWER switch of this unit does not control the power supply to connected equipment.)

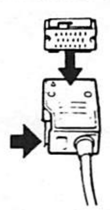
❼ ⏚ ground terminal
To reduce hum in the audio signal, connect this terminal to a ground terminal of the connected audio equipment.



❽ REMOTE connector (33-pin)
Connect an optional editing control unit, auto search control unit or remote control unit to this connector.
- Before connecting the remote control cable, check whether the connector is male or female.
- The REMOTE connector accepts a 20-pin connector. A plug adaptor is unnecessary.



To disconnect the cable, press the white button on the connector and pull the connector out.



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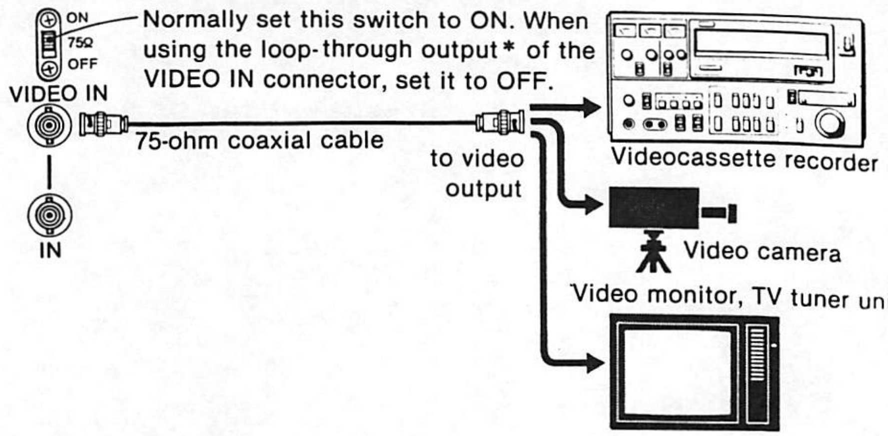
⑨ VIDEO IN connectors (BNC type) **and 75Ω termination switch**

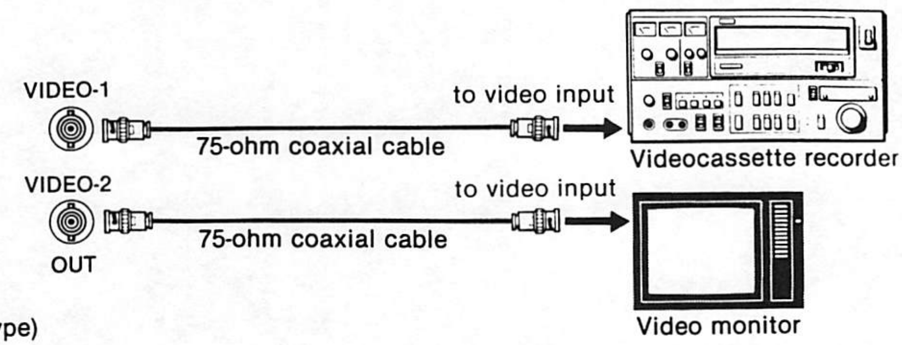
Connect the video signal to be recorded to one of these connectors.

*These two connectors are in "loop-through" configuration, so that the input signal from one connector is fed directly to the other connector. Use this loop-through output as the video input source for other video equipment.



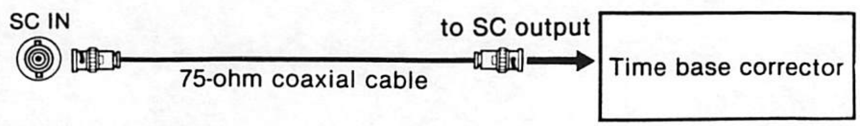
⑩ VIDEO OUT connectors (BNC type)

The video signals are simultaneously output at these connectors.



⑪ SC IN (subcarrier input) connector (BNC type)

Connect the subcarrier from the time base corrector here.

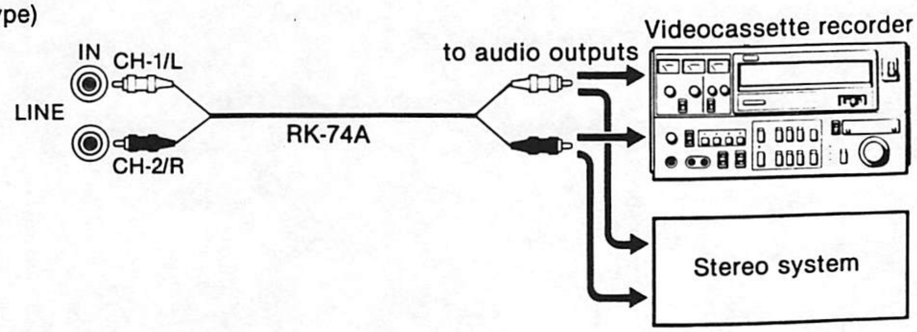


⑫ COLOR LOCK switch

As a rule, set to NORMAL. If the playback picture has no color or if the hue is abnormal set the switch to the position marked [•].

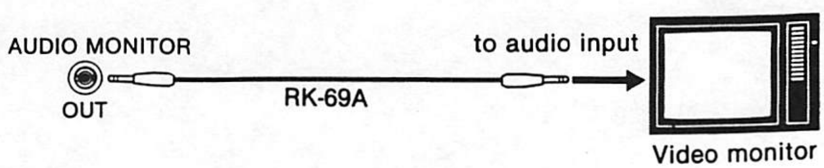
⑬ LINE IN (audio line input) jacks (phono type)

Connect the audio signal to be recorded here.

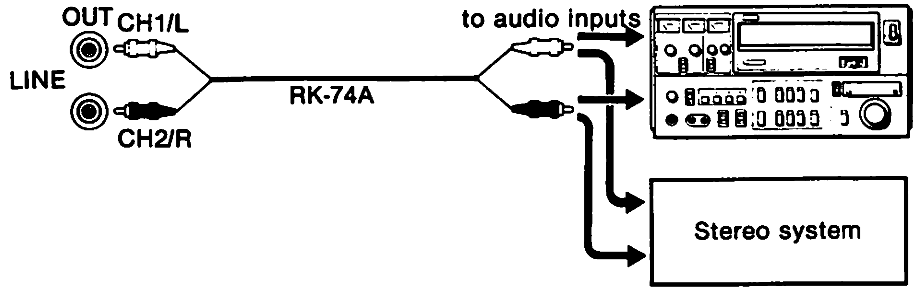


⑭ AUDIO MONITOR jack (mini type)

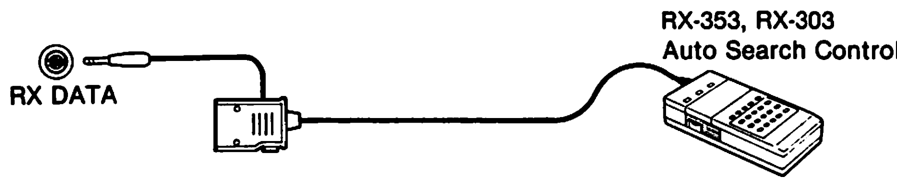
Connect to the audio input jack on the video monitor. The signal selected by the AUDIO MONITOR switch on the front panel is output here.



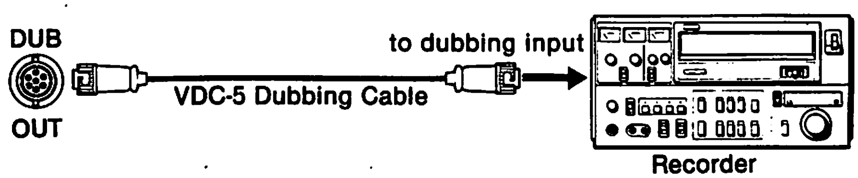
⑮ **LINE OUT (audio line output) jacks** (phono type)
The signals recorded on the audio channel 1 and audio channel 2 are output here.



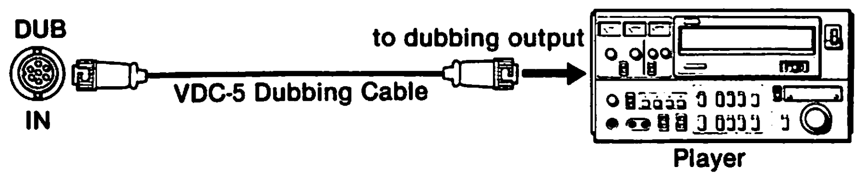
⑯ **RX DATA jack** (mini type)
For recording and reading the data recorded on the tape by the RX-353.



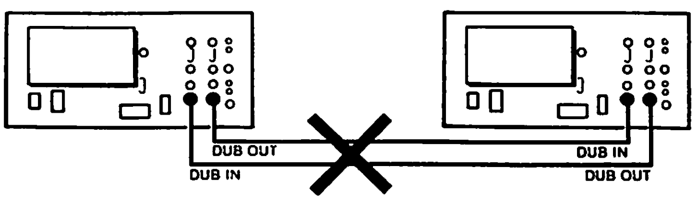
⑰ **DUB OUT connector** (7 pin)
When duplicating a tape using a recorder with a dubbing connector, the video signal is connected using this connector.

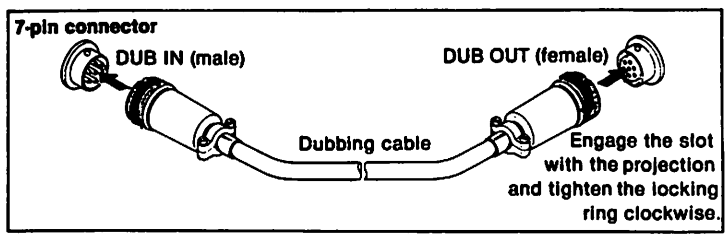


⑱ **DUB IN connector** (7 pin)
When duplicating a tape using a player with a dubbing connector, the video signal is connected using this connector.



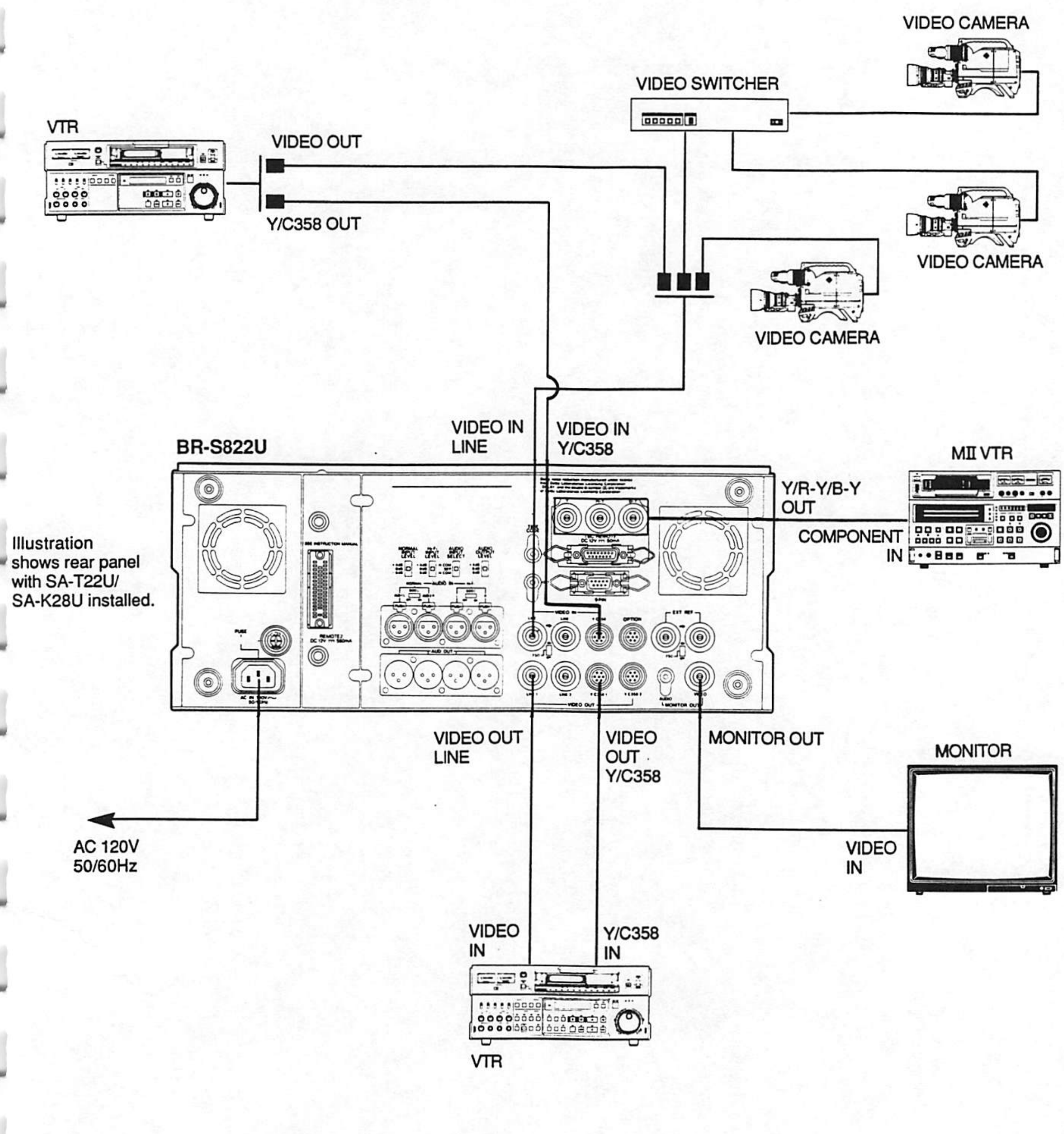
Note
Do not connect the dubbing connectors in parallel.





CONNECTIONS

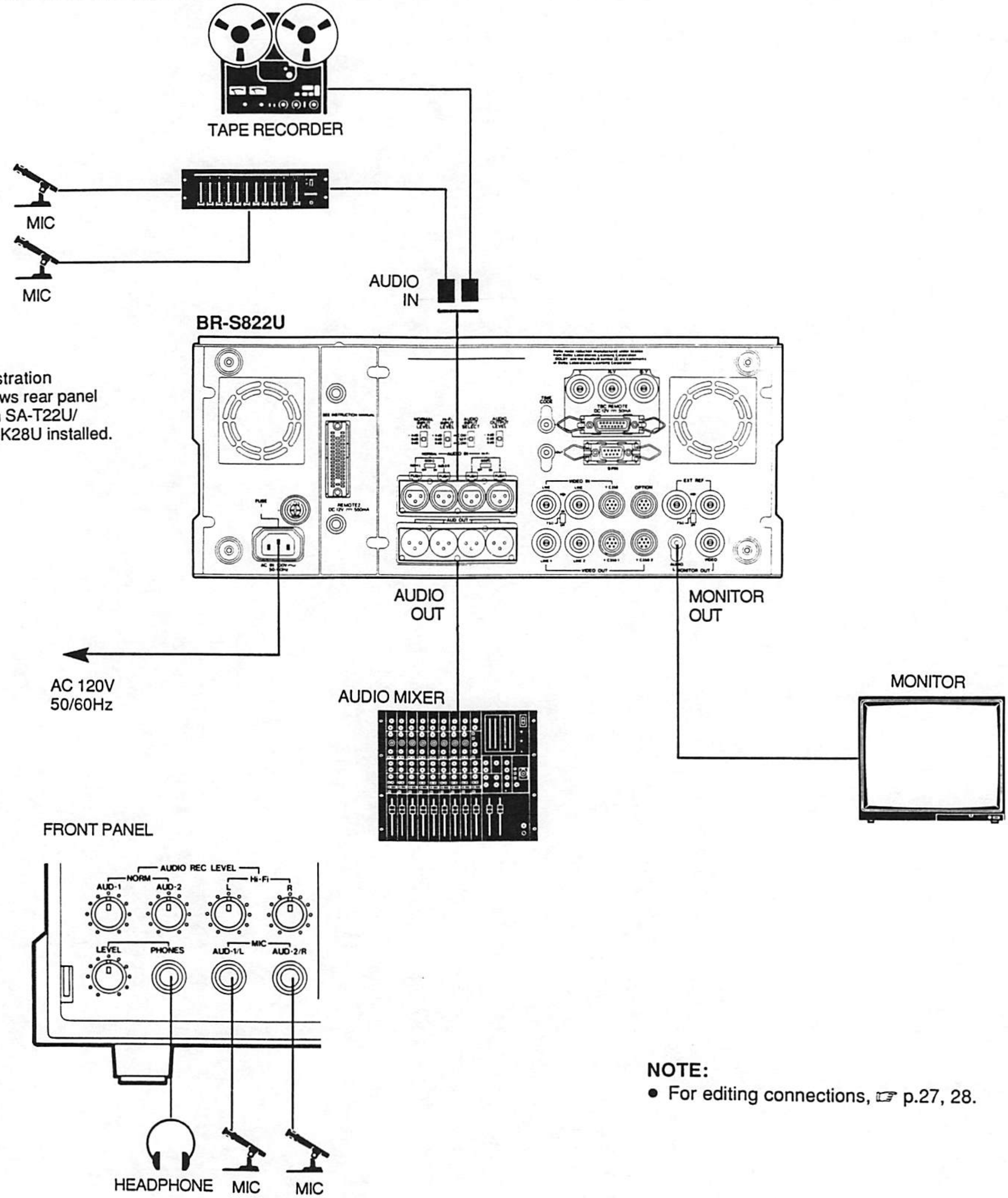
VIDEO EQUIPMENT



OTES:

- To output the loop-through signal, set the 75-ohm terminating switch to OFF, otherwise set it to ON. (Be sure to terminate the signal at the last of the connected units.)
  On-screen information is output from the VIDEO MONITOR OUT connector only.
- Y/R-Y/B-Y component signals can be output when the optional TBC board SA-T22U is installed. M-II and Betacam component signals are selectable via menu item #104. (☞ p.39)

AUDIO EQUIPMENT



- Illustration shows rear panel with SA-T22U/ SA-K28U installed.

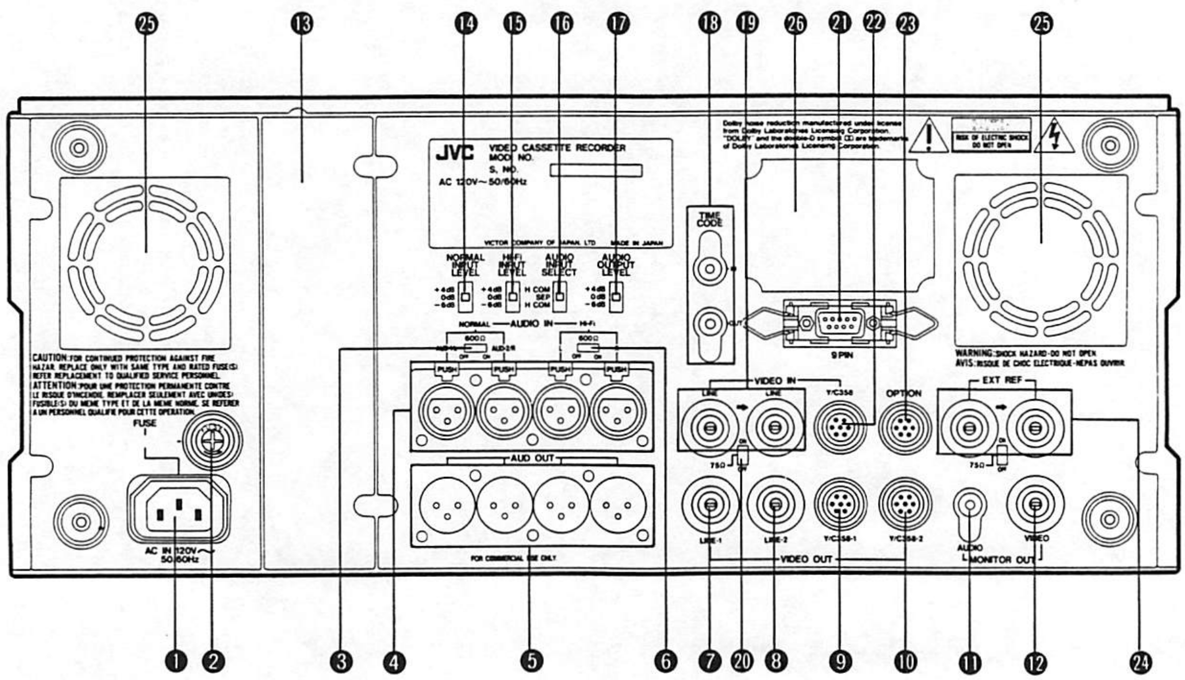
NOTE:

- For editing connections, ☞ p.27, 28.

NOTES:

- The MIC jack has priority over the rear panel AUDIO IN connectors. When a microphone is connected, the input signal is automatically shifted from AUDIO IN to MIC.

REAR PANEL



❶ **AC IN socket**
- Connect to 120 V AC, 50/60 Hz power outlet.

❷ **Fuse holder**

❸ **NORM AUDIO INPUT impedance select switch**

ON: 600 ohms.
OFF: 10 k-ohms. Normally set to this position.

❹ **Audio input connectors**

AUDIO IN NORMAL: Normal audio input connectors for Audio-1 and Audio-2.
AUDIO IN Hi-Fi: Hi-Fi audio input connectors for Left and Right.

❺ **Audio output connectors**

AUDIO OUT NORMAL: Normal audio output connectors for Audio-1 and Audio-2.
AUDIO OUT Hi-Fi: Hi-Fi audio output connectors for Left and Right.

❻ **Hi-Fi AUDIO INPUT impedance select switch**

ON: 600 ohms.
OFF: 10 k-ohms. Normally set to this position.

❼, ❽ **VIDEO OUT LINE (1, 2) connectors**
- The composite video signal is output from these connectors.

❾, ❿ **VIDEO OUT Y/C358 (1, 2) connectors**
- The Y/C358 signal is output from these connectors.

⓫ **AUDIO MONITOR OUT connector**
- The audio signal selected with the AUDIO MONITOR select switches is available at this connector.

⓬ **VIDEO MONITOR OUT connector**
- The composite video output signal is available at this connector. On-screen information is also supplied.

⓭ **Expansion slot**
- For installation of optional interface (SA-K28U or SA-K27U).

⓮ **NORMAL INPUT LEVEL select switch**
- To select -6 dB, 0 dB, or +4 dB according to the level of the normal audio input signal. Both channels are switched simultaneously.

⓯ **Hi-Fi INPUT LEVEL select switch**
- To select -6 dB, 0 dB, or +4 dB according to the level of the Hi-Fi audio input signal. Both channels are switched simultaneously.

⓰ **AUDIO INPUT SELECT switch**

H COM: "Hi-Fi Combined" recording. Set to this position to record audio signals input to the AUDIO IN Hi-Fi connectors on both the Hi-Fi and Normal audio tracks.
SEP: "Separate" recording. Set to this position to record audio signals input to the AUDIO IN Hi-Fi and NORMAL connectors separately on the Hi-Fi and Normal audio tracks.
N COM: "Normal Combined" recording. Set to this position to record audio signals input to the AUDIO IN NORMAL connectors on both the Hi-Fi and Normal audio tracks.

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17 **AUDIO OUTPUT LEVEL select switch**
- To select -6 dB, 0 dB, or +4 dB according to the input level of connected audio equipment. All four audio channels are switched simultaneously.

18 **TIME CODE IN/OUT connectors**
Set menu item #206 to "01 – LTC" to record LTC time codes on the normal audio-2 track.
- Connect a time code generator to the IN connector for external time code recording.
- Connect a time code reader to the OUT connector for external time code reading.

19 **VIDEO IN LINE connectors**
- The composite video signal is input to the left connector.
- To output the loop-through signal to another unit, set the 75-ohm terminating switch to OFF.

20 **75-Ohm terminating switch**
ON: The loop-through signal is terminated at the BR-S822U.
OFF: The loop-through signal is output to another unit.

21 **9-PIN connector**
- Connect to an RS-422 9-pin serial remote control unit or to the RS-422 9-pin connector of a feeder for swap editing.

22 **VIDEO IN Y/C358 connector**
- The Y/C358 signal is input to this connector.

23 **OPTION connector**
- Delivers the Y/C688 signal (with optional SA-E68U Output board installed) to the DUB IN connector of 3/4" U-VCR machines.

24 **EXT REF connectors with 75-ohm terminating switch**
- Supply the reference signal (either black burst signal or composite video) to the left connector and set the 75-ohm terminating switch to ON.
- To output a loop-through signal to another unit, set the 75-ohm terminating switch to OFF.

NOTE:
- When using the SA-T22U, do not use a black-and-white signal or sync signal without burst as the reference signal, otherwise the intended synchronization will not be obtained.

25 **Fans**

26 **Expansion slot**
For installation of COMPONENT OUT connector board when optional SA-T22U TBC is installed.

2-4. Connector Panel

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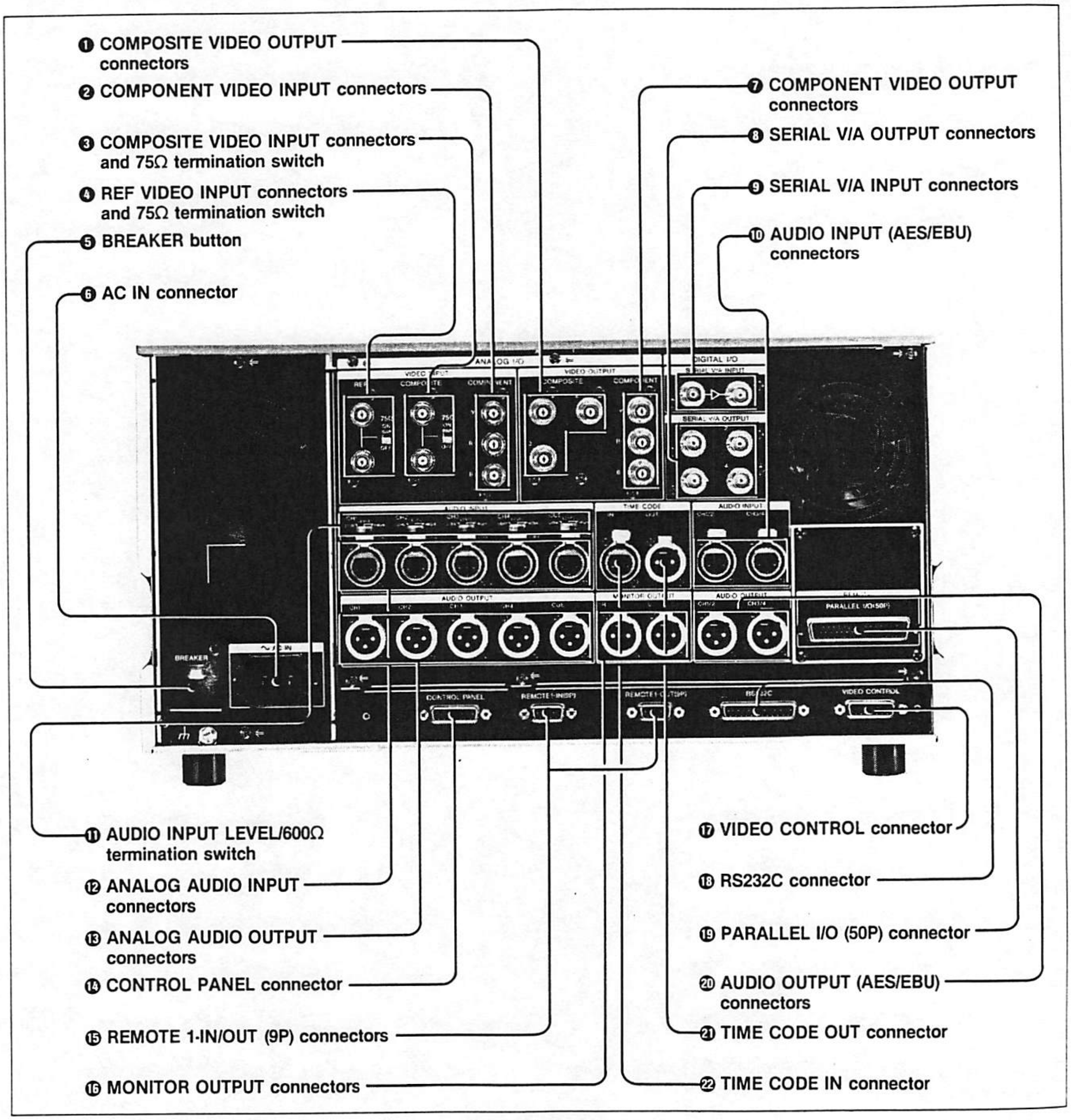


Connector panel

❶ COMPOSITE VIDEO OUTPUT connectors (BNC)
Output analog composite video signals. The signal output from connector 3 (SUPER) contains superimposed characters for time data or menu data when the CHARACTER switch is set to ON.

❷ COMPONENT VIDEO INPUT connectors (BNC)
Accept analog component video signals (Y/R-Y/B-Y).

❸ COMPOSITE VIDEO INPUT connectors (BNC) and 75Ω termination switch (with optional BKDW-505 for NTSC system, or BKDW-506 for PAL system)
Accept an analog composite video signal.
Set the 75Ω termination switch to OFF when using both of the COMPOSITE VIDEO INPUT connectors for a bridging connection. Otherwise, set it to ON.

❹ REF. (reference) VIDEO INPUT connectors (BNC) and 75Ω termination switch
One of these connectors accepts a reference video signal. Use an analog video signal with chroma burst (VBS) or a black and white video signal (VS) as a reference video signal.
When making a bridging connection with a loop-through output, set the 75Ω termination switch to OFF. Otherwise, set it to ON.

❺ BREAKER button
Disconnects the primary circuit of the AC power transformer should an excessive current be defected.

❻ AC IN connector
Connect to an AC outlet using the power cord supplied with the VTR.

❼ COMPONENT VIDEO OUTPUT connectors (BNC)
Output analog component video signals (Y/R-Y/B-Y).

❽ SERIAL V/A (video/audio) OUTPUT connectors (BNC)
Output four (1 to 4) serial digital video/audio signal lines. The signal output from connector 4 (SUPER) contains superimposed characters for time data or menu data when the CHARACTER switch is set to ON.

❾ SERIAL V/A (video/audio) INPUT connectors (BNC)
The left connector accepts serial digital video/audio signals. When the VTR is powered on, the right connector serves as an active loop-through output.

❿ AUDIO INPUT (AES/EBU) connectors (XLR-3-31)
Accepts a maximum of two lines (four channels: channels 1/2 and channels 3/4) of AES/EBU format digital audio signals.

⓫ AUDIO INPUT LEVEL/600Ω termination switch
Set according to the audio input level of each channel input to the ANALOG AUDIO INPUT connectors and for the audio input impedance.

LOW with OFF
Audio input level: −60 dBu (microphone input)
Audio input impedance: High (about 20 kilohms)

HIGH with OFF
Audio input level: +4 dBu (line input)
Audio input impedance: High (about 20 kilohms)

HIGH with ON
Audio input level: +4 dBm (line input)
Audio input impedance: 600Ω

⓬ ANALOG AUDIO INPUT connectors (XLR-3-31)
Accept up to five analog audio signal lines (channels 1 to 4 and cue).

⑬ ANALOG AUDIO OUTPUT connectors (XLR-3-32)
Output up to five analog audio signal lines (channel 1 to 4 and cue).

⑭ CONTROL PANEL connector (15-pin)
Connects the control panel using the 15-pin cable supplied with the optional BKDW-510 Control Panel Extension Kit when using the control panel as a remote controller.

⑮ REMOTE 1-IN (9P)/OUT (9P) connectors (D-sub 9-pin)
Connect to another DVW-A500/500 series VTR or D-1, D-2, or Betacam SP VTR via the 9-pin remote control cable supplied with the VTR.
Used when you edit using two VTRs and the BVE-900/910/2000/9000/9100 editing control unit. The REMOTE 1-IN and OUT connectors can be used to make a bridge connection.

⑯ MONITOR OUTPUT connectors (XLR-3-32)
Output signals to the audio monitor. These connectors output two signal lines: L and R.
Select the signals to be output with the MONITOR SELECT button and the AUDIO INPUT/MONITOR SELECT buttons.
You can make setting to enable adjustment of the volume level with the PHONES level control of the upper control panel.

Refer to "1-9. Switch Settings on the Connector Panel and Boards" in the Installation Manual.

⑰ VIDEO CONTROL connector (D-sub 15-pin)
Connects to the optional BVR-50/50P remote control unit to enable remote control of the video processor.
Before connecting the remote control unit, turn off the power of the VTR.

⑱ RS232C connector (D-sub 25-pin)
Accepts or sends the RS-232C remote control signal and/or VTR status data from/to the external equipment. When this connector is being used for communication, the RS-232C indicator on the upper control panel lights.

⑲ PARALLEL I/O (50P) connector (D-sub 50-pin, with optional BKDW-509)
Inputs an external remote control signal.

For details, refer to the Installation Manual.

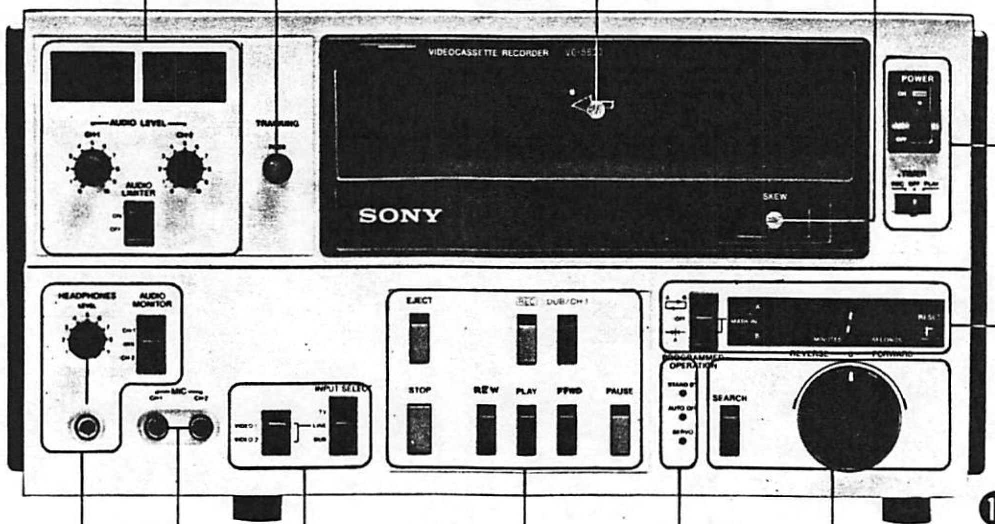
⑳ AUDIO OUTPUT (AES/EBU) connectors (XLR-3-32)
Output a maximum of two lines (four channels: channels 1/2 and 3/4) of AES/EBU format digital audio signals.

㉑ TIME CODE OUT connector (XLR-3-32)
Outputs one of the following time codes according to the VTR operation mode.
In playback mode: Playback time code.
In record mode: Time code generated by the internal time code generator, or that input to the TIME CODE IN connector.

㉒ TIME CODE IN connector (XLR-3-31)
When recording a time code supplied from external equipment, this connector accepts the time code signal.
Connect to the time code output terminal of the external equipment.

LOCATION AND FUNCTION OF CONTROLS

FRONT PANEL

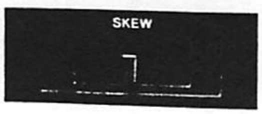


❶ Power section

POWER switch
Press the side with the dot to turn the recorder on or to operate the recorder using the timer. To turn the recorder off, set the switch to OFF.

TIMER switch
Timer-activated recording or playback is controlled with this switch when the POWER switch is set to ON.
REC : To start and stop recording using the timer
OFF : To turn the timer off.
PLAY : To start and stop playback using the timer
- When the switch is set to REC, the function buttons other than the STOP button are not activated.

❷ SKEW control



This control adjusts tension on the tape. Slide the control to the right or the left to obtain a normal picture when picture "hooking" distortion appears in the upper part of the screen. Normally, set the control to the center detent position. This control returns automatically to the center position when the unit is set to the record mode.
- Do not move this control during recording.

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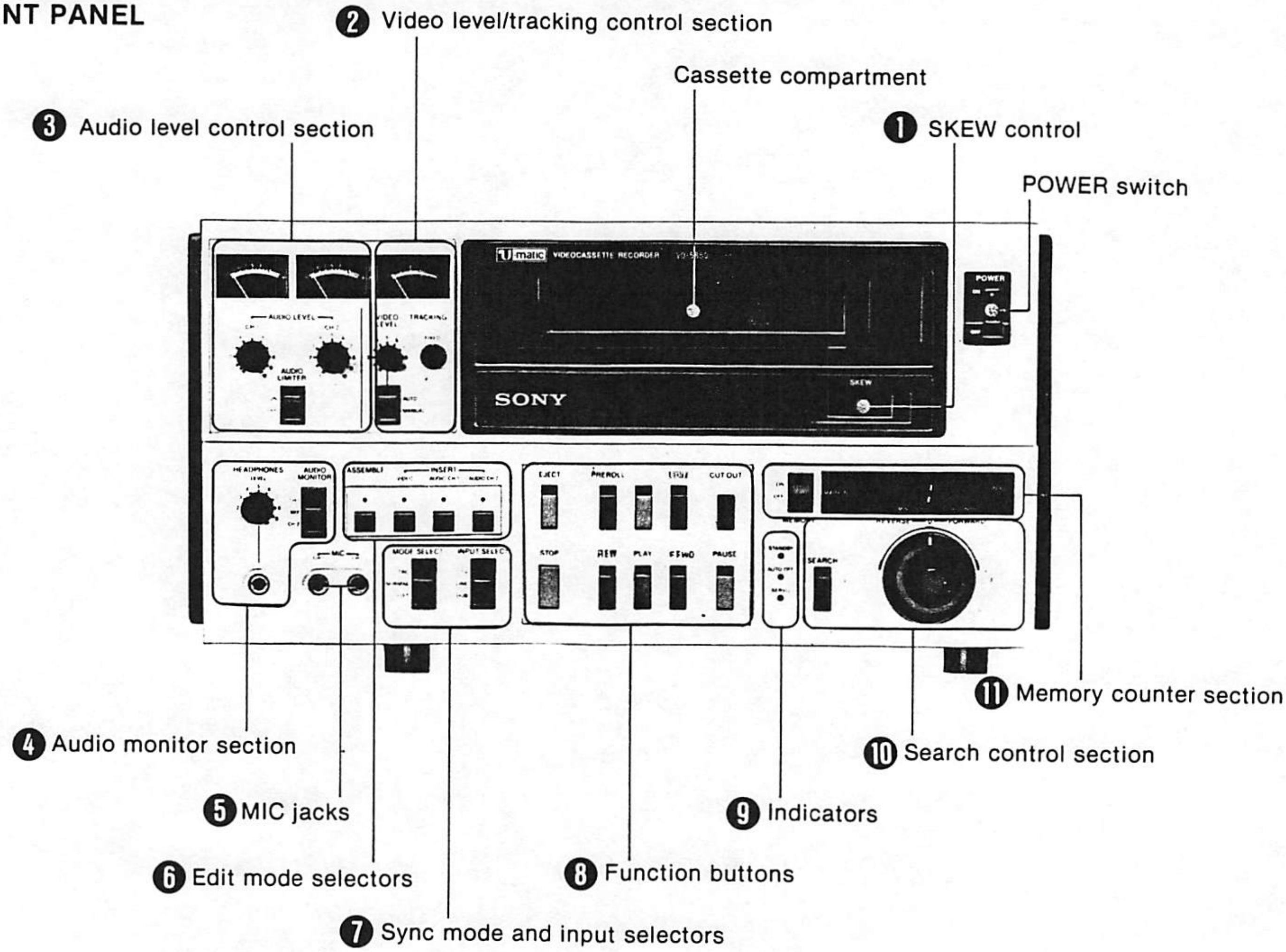
LOCATION AND FUNCTION OF CONTROLS

FRONT PANEL



❶ SKEW control

This control adjusts tension on the tape. Slide the control to the right or the left to obtain a normal picture when picture "hooking" distortion appears in the upper part of the screen. Normally, set the control to the center detent position. This control returns automatically to the center position when the unit is set to the record or edit mode.

- Do not move this control during recording or editing.

❷ Video level/tracking control section

VIDEO LEVEL/TRACKING meter
This meter indicates the video recording level during recording and whether or not tracking is correct during playback.

TRACKING control
Turn to minimize the tracking variance between machines so that the tracking meter pointer deflects to the right as far as possible.
Normally, set this control to the center FIXED position.

VIDEO LEVEL control
With the AUTO/MANUAL switch set to MANUAL, turn this control to adjust the video recording level so that the meter pointer swings within the blue zone.

AUTO/MANUAL switch
AUTO : The video recording level will be adjusted automatically.
MANUAL : Set to this position to adjust the video recording level manually.

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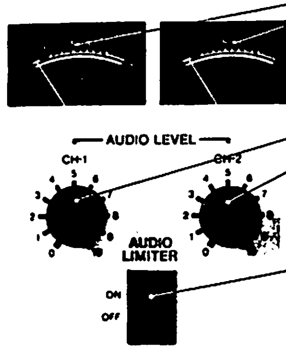
❸ Audio level control section

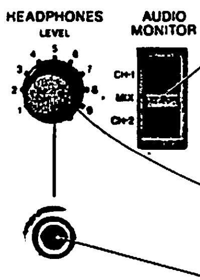


AUDIO LEVEL meters
These meters indicate the audio recording level during recording and the audio playback level during playback.

AUDIO LEVEL controls
These controls adjust the audio recording level of the channel 1 or 2 separately. Turn the controls so that the meter pointers swing up to "0" at their maximum deflection.

AUDIO LIMITER switch
Set the switch to ON when recording a program which contains peaks such as live recording. The audio limiter circuitry will be activated to minimize audio distortion at the peaks. For normal recording and when adjusting the audio recording level, set the switch to OFF.

❹ Audio monitor section

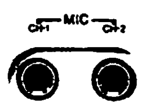


AUDIO MONITOR switch
This switch selects the monitoring sound through a video monitor or headphones.
CH-1 : To hear the sound recorded on channel 1 only.
MIX : To hear the channel 1 sound from the left headphone and the channel 2 sound from the right headphone or to hear the mixed sound of the channel 1 and channel 2 through the video monitor connected to the AUDIO MONITOR jack or TV connector.
CH-2 : To hear the sound recorded on channel 2 only.
●This switch does not affect the audio outputs of the LINE OUT jacks.

HEADPHONES LEVEL control
This control adjusts the volume at the headphones.

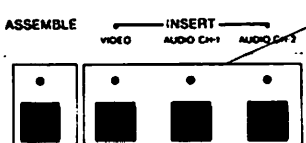
HEADPHONES jack (stereo phone jack)

❺ MIC jacks (phone jacks)



For microphone recording, connect microphones with phone plugs here. The CH-1 jack is for the left microphone and the CH-2 jack is for the right microphone. When a microphone is connected, the corresponding LINE IN jack will be automatically cut off.
To connect microphones with mini plugs, use the optional PC-2A plug adaptors.

❻ Edit mode selectors



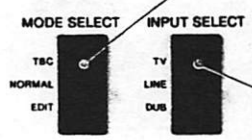
INSERT buttons and indicators
These buttons select the signal(s) to be inserted when editing. The indicator(s) corresponding to the button(s) pressed will light.
VIDEO : To insert the video signal.
AUDIO CH-1 : To insert the sound on audio channel 1.
AUDIO CH-2 : To insert the sound on audio channel 2.
●Two or more buttons can be pressed simultaneously.

ASSEMBLE button and indicator
Press for assembly editing. The indicator will light.

To cancel the mode selected by one of these buttons, press the button again. The indicator will go out.

To change the mode from assembly to insert or vice versa, first cancel the previously selected mode then press the appropriate buttons.

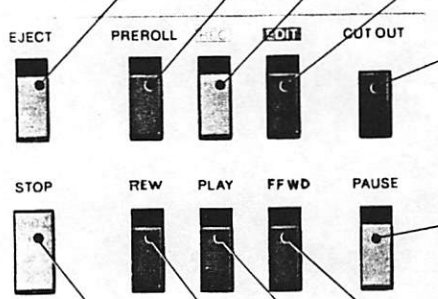
❼ Sync mode and input selectors



MODE SELECT switch
This switch selects the sync mode according to the connected equipment or to the operating mode.
TBC: For operation with a time base corrector.
NORMAL: For normal recording or playback.
EDIT: For editing.
● If the switch is set to TBC without connecting a time base corrector, the vertical sync does not lock in the search mode and the picture rolls vertically.

INPUT SELECT switch
This switch selects the signal to be recorded.
TV: To record the signal from the TV connector.
LINE: To record the signal from the VIDEO IN connector.
DUB: To record the signal from the DUB IN connector.

❽ Function buttons



EJECT button
Press to eject the video cassette.

PREROLL button and indicator
Press this button in pause mode and the tape rewinds by 5 seconds and stops in pause mode.

REC (record) button and indicator
For recording, press this button simultaneously with the PLAY button. While this button is pressed, the E-to-E mode picture* can be seen.

EDIT button and indicator
For editing, press this button simultaneously with the PLAY button. While this button is pressed, the E-to-E mode picture* can be seen.

CUT OUT button
Press this button when assembly or insert editing is finished. The edit mode is cancelled and the tape will continue to run in playback mode. When this button is pressed in record mode, the record mode is cancelled, and when this button is pressed in pause mode after the preroll, the edit mode is cancelled.

PAUSE button and indicator
Press to stop the tape momentarily. To start the tape, press again. When this button is pressed during playback, a still picture will be obtained. If the PLAY, F FWD, REW or SEARCH button is pressed during the pause mode, the pause mode will be released and the tape will run in the mode designated by the button pressed.

F FWD (fast forward) button and indicator
Press to advance the tape rapidly. The E-to-E mode picture* can be seen on the monitor screen.

PLAY button and indicator
Press to play the tape back. Simultaneously pressing this button with the REC button sets the unit in the record mode; simultaneously pressing it with the EDIT button sets the unit in the edit mode.

REW (rewind) button and indicator
Press to rewind the tape. The E-to-E mode picture* can be seen on the monitor screen.

STOP button
Press to stop the operation of the unit. The E-to-E mode picture* can be seen on the monitor screen.

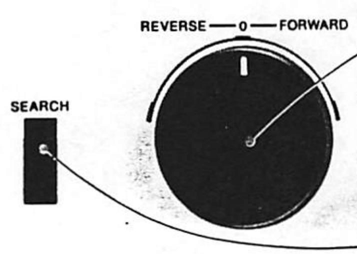
⑨ Indicators

STAND BY indicator
This indicator lights while the tape is being threaded or unthreaded.

AUTO OFF indicator
This indicator lights to indicate that the moisture is condensed inside the unit. While the AUTO OFF indicator is lighting, the unit does not operate even if the function button is pressed.

SERVO indicator
This indicator lights when the drum servo and capstan servo are locked and the tape transport mechanism is stabilized. When the tape just begins to move, this indicator may flicker.

⑩ Search control section



Search dial
After pressing the SEARCH button, turn this dial to control the speed and direction of tape travel during the search for a particular point on the tape. You can vary the tape speed from 1/30 to 5 times normal playback speed. When the dial is set at the center "0" position, the picture will be still.
● When this dial is used, the servo system is not locked so the guard band noise flows on the playback picture.

SEARCH button
Press to engage the search dial.

⑪ Memory counter section

RESET button
Press to reset the time counter to "00 minutes 00 seconds". The point memorized by the MARK IN button will be cancelled.

Time counter
This counter counts the CTL signals on the tape during playback, and the CTL signals being recorded during recording and indicates the tape running time at the normal playback speed in minutes and seconds (up to 99 minutes 59 seconds).

MARK IN button
Press to memorize a particular point on the tape and the time indicator keeps indicating the memorized point while the button is pressed.
The memory function can be activated at any position of the MEMORY switch.

MEMORY switch
When this switch is set to ON, the tape will automatically stop the point where the MARK IN button was pressed or, if the MARK IN button has not been pressed, where the time counter indicates "00 00" in the rewind or search mode.
● During editing or when the REMOTE connector is used, be sure to set this switch to OFF.

* E-to-E mode picture
The input video signal which is frequency-modulated and then demodulated in the recorder, is displayed on the monitor. This is the E-to-E (Electronics to Electronics) mode picture.

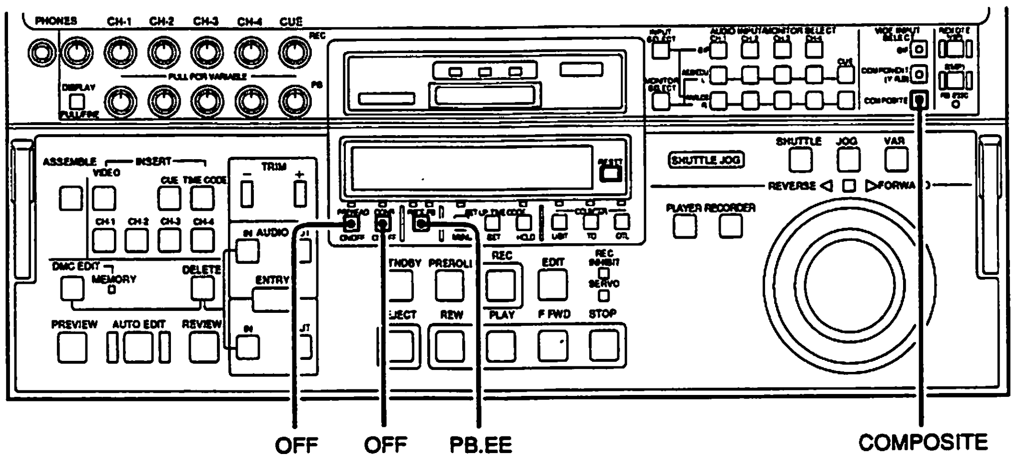
After installing the board, turn on the power. Wait more than 30 minutes after turning on the power, then perform the adjustment.

[Equipment]

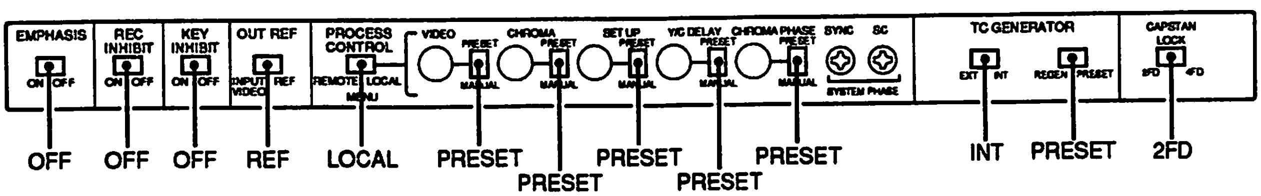
•Analog Composite Signal Generator (TEKTRONIX TSG-170A or equivalent)
•Waveform Monitor (TEKTRONIX 1750, 1780R or equivalent)

[Switch Setting]

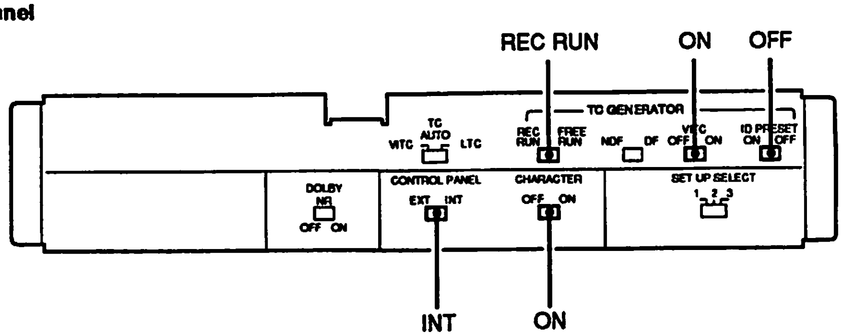
Upper/Lower Control Panel



Sub Control Panel



System Setup Panel

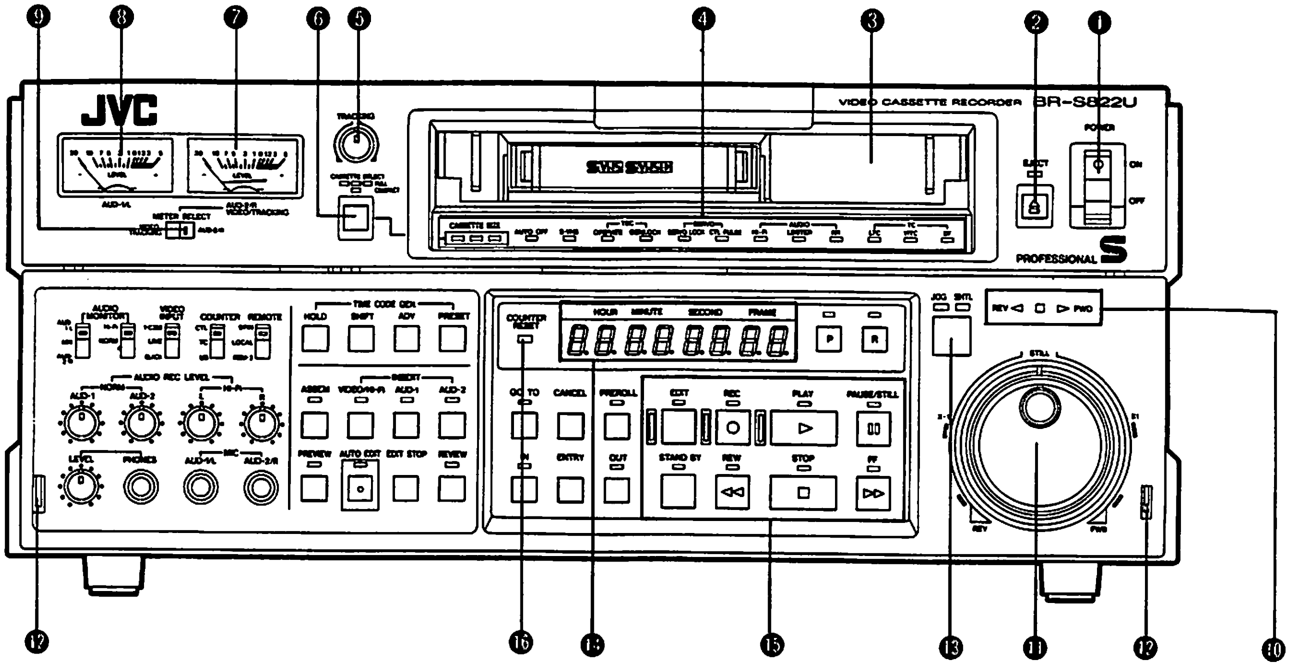


SS-52 Board

Set the switch S100-1 to ON (CLOSE).

CONTROLS AND CONNECTORS

FRONT PANEL



❶ **POWER switch**
- When power is ON, the time counter and level meters will be illuminated.

❷ **EJECT button with LED indicator**
- Ejects the cassette (from any mode).
- The indicator lights while the cassette is being ejected.

❸ **Cassette loading slot**
- Accepts either a compact or full-size S-VHS/VHS cassette according to the type selected with the CASSETTE SELECT button ❻.

❹ **LED indicators**

CASSETTE SIZE indicators
- Indicate whether the recorder is in the Full or Compact mode. When all three indicators are blinking, the recorder is ready to accept a full-size cassette. When only the center indicator is blinking, the recorder is ready to accept a compact cassette. Press the CASSETTE SELECT button ❻ to change modes. When a cassette is inserted, the blinking will stop and the corresponding indicator(s) will remain continuously lit.

AUTO OFF indicator
- Lights when the unit malfunctions. All other controls are disabled.

S-VHS indicator
- Lights when an S-VHS or S-VHS-C cassette is inserted with the unit in the S-VHS mode, or when playing back a blank part of the tape.
- Blinks when S-VHS recording is attempted with a VHS cassette.

TBC indicators
(with optional SA-T22U TBC installed)

OPERATE: Lights when the TBC is in operation. A timebase-corrected signal is output.

GENLOCK: Lights when the TBC is in operation and locked to the external reference signal.

SERVO indicators

SERVO LOCK: Lights when the capstan and drum servos are locked to the reference signal.

CTL PULSE: Lights during playback of a tape with no control pulse recorded.

AUDIO indicators

Hi-Fi: Lights when the Hi-Fi REC circuit is ON (via menu setting) or when playing back Hi-Fi-recorded tapes.

LIMITER: Lights when the built-in audio limiter circuit is set to ON (via menu setting).

NR: Lights when the Dolby B* noise reduction circuit is set to ON (via menu setting).

TC (TIME CODE) indicators
(with optional SA-R22U TC [time code] generator/reader installed)

LTC: Lights green when LTC-recorded tapes are played back with the normal audio-2 track set for LTC use (via menu setting). If LTC is not picked up, the indicator lights orange. This indicator may also light green when normal-audio-recorded tapes are played back.

VITC: Lights when VITC-recorded tapes are played back or when recording a VITC signal.

DF: Lights when recording or playing back in the Drop-Frame mode.

❺ **TRACKING control**
- Adjusts tracking. Turn in either direction until the tracking meter deflects all the way to the right.
- Normally leave in the center click-stop position.

9 2

6 CASSETTE SELECT button
- Press to select FULL or COMPACT. The corresponding indicator(s) will light.

7 AUD-2/R (VIDEO/TRACKING) level meter
- Indicates the audio level of the normal audio-2 or Hi-Fi right-channel signal during recording and playback.
- Functions as a video level meter during recording and as a tracking meter during playback when the METER SELECT switch 9 is set to VIDEO/TRACKING.

8 AUD-1/L level meter
- Indicates the audio level of the normal audio-1 or Hi-Fi left-channel signal during recording and playback.

9 METER SELECT switch
- Switches the AUD-2/R level meter 7 between audio level and video level indication.

AUD-2/R:	Meter functions as the audio-2/Hi-Fi right-channel level meter.
VIDEO/TRACKING:	Meter functions as a video level meter in recording, and as a tracking meter in playback.

10 Tape direction indicators
- Indicate the current tape direction.

▷: Forward
□: Still
◁: Reverse

11 JOG/SHUTTLE dials
- Dual concentric controls. The outer functions as a Shuttle ring, the inner as a Jog dial. The Jog and Shuttle modes can be entered directly from the Play, Still, FF, REW, or Stop modes.

SHUTTLE ring:	Search speed can be varied continuously from 1/30 to 32 times normal (up to 10 times normal with C-size cassettes) in forward or reverse. Set to the center click-stop position to engage the Still mode.
JOG dial:	Manual frame-by-frame search in either direction. Tape speed is determined by the speed of dial rotation. Releasing the dial engages the Still mode. Also used in edit point trimming, menu setting and TC/UB presetting.

12 Control panel lock release buttons
- To tilt the control panel, press these buttons and lift the panel at the same time. The panel can be tilted to 90° and locked at angles of 25°, 50°, and 75°.

13 JOG/SHUTTLE button with JOG/SHTL mode indicators
- Instantly re-activates the Shuttle mode with search speed determined by the current dial setting.

14 Time counter
- Shows tape time in hours, minutes, seconds, and frames.
- Displays edit-in and -out points.
- Displays user bits.
- Displays menu settings and warnings.

15 Operation buttons with LED indicators

PAUSE/STILL button
- Temporarily stops recording when pressed in the Record mode.
- Displays a still picture when pressed in the Play mode.

PLAY button
- Starts playback.
- Re-starts normal playback when pressed in the Still or Search mode.
- Starts recording when pressed together with the REC button.
- Starts editing when pressed together with the EDIT button in the Play mode (Run Editing).
- Re-starts recording when pressed in the Record-Pause mode.

REC button
- Starts recording when pressed together with the PLAY button.
- Outputs EE signals when pressed in the Play mode.
- Displays TC generator data when pressed in the Stop mode. (Released by pressing STOP button.)

EDIT button
- Starts editing when pressed together with the PLAY button in the Play mode.
- Outputs EE signals (selected with the Edit Mode Select buttons) when pressed on its own in the Play mode.
- Displays TC generator data when pressed in the Stop mode. (Released by pressing STOP button.)

STAND BY button
- Switches the recorder between the Standby-On and Standby-Off modes while the VCR is in the Stop mode. Standby-On is automatically engaged when the Stop button is pressed.

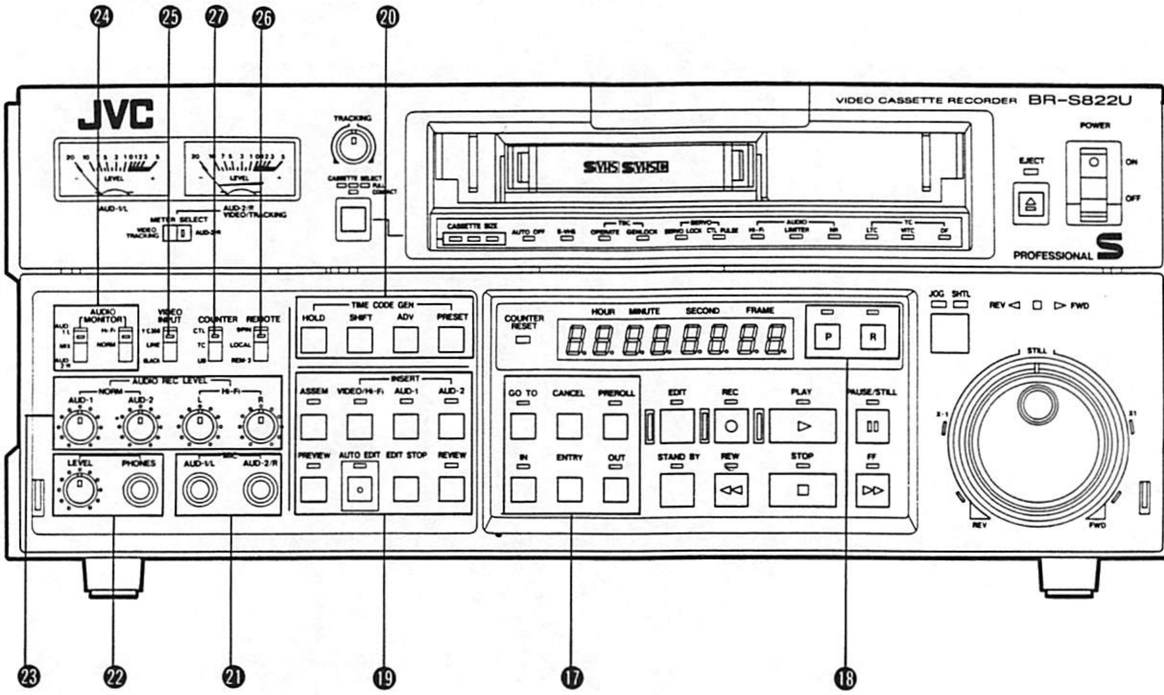
Standby-On:	The tape is loaded and the drum is rotating. The indicator is lit.
Standby-Off:	The tape is loaded but tape tension is reduced and the drum does not rotate. The indicator is not lit.

REW button
- Starts rewind when pressed in any mode.

STOP button
- Engages the Stop mode (Standby-On). The tape stops, but remains in the full-loaded position with the drum rotating.
- The STOP and STAND BY indicators will light.

FF button
- Starts fast forward when pressed in any mode.

16 COUNTER RESET button
- Resets the CTL counter to zero.
- Clears the entered edit point.
- The CTL counter will be reset even if this button is pressed in the TC mode.



17 Edit control buttons

PREROLL button with LED indicator

- Prerolls the tape by about 5 seconds.

CANCEL button

- Press together with the IN or OUT button to clear the edit point from memory.

GO TO button

- Press together with the IN or OUT button to access the IN or OUT point.

ENTRY button

- Press together with the IN or OUT button to enter an IN or OUT point.

IN/OUT buttons with LED indicators

- Press together with the ENTRY button to enter the IN or OUT point.
- Press either button on its own to display the IN or OUT point.
- Press simultaneously to display edit duration.
- Turn the JOG dial while holding either button to trim the IN or OUT point.

18 Player/Recorder select buttons

For swap editing via the 9-pin connector.

- Press P to operate the Player with this recorder's controls.
- Press R to operate this recorder.

19 Edit operation buttons with LED indicators

Edit mode select buttons

- To select the editing mode.

ASSEM:	All input video and audio signals are recorded.
VIDEO/HI-FI:	Inserts the video signal and the Hi-Fi audio signal together.
AUD-1:	Inserts the normal audio-1 signal.
AUD-2:	Inserts the normal audio-2 signal or the LTC signal.

REVIEW button

- Reviews the executed edit.

EDIT STOP button

- Stops automatic editing.

AUTO EDIT button

- Executes automatic editing.

PREVIEW button

- Previews the programmed edit.

20 Time Code setting buttons

To preset time code/user bit data (with optional SA-R22U TC generator/reader installed).

HOLD button

This button is only effective when the SA-R22U's PRESET/REGEN switch is set to PRESET.

- Holds the current counter data; the leftmost digit will blink.

SHIFT button

- Shifts the blinking digit to the right. (You can also shift the blinking digit in either direction by holding down the SHIFT button and turning the JOG dial.)

ADV (ADVANCE) button

- Advances the value of the blinking digit. (You can also change the value in either direction by holding down the ADV button and turning the JOG dial.)

PRESET button

- Transfers the data set with the HOLD, SHIFT, and ADV buttons to the time code generator.
- Automatically cancels the Hold mode.

21 MIC jacks (AUD-1/L, AUD-2/R)

- For microphone connection. Input signal switches from line to microphone.

㉒ PHONES jack/LEVEL control
- Connect a set of headphones to monitor sound recording.
- Adjust output level with the LEVEL control.

㉓ Hi-Fi L/R and NORM AUD-1/AUD-2 AUDIO REC LEVEL controls
- To separately adjust recording levels for the Hi-Fi left/right-channel signals and the normal (linear) audio-1/2 signals.
- Optimum level is the point where the corresponding meter's peak deflection is "0".

㉔ AUDIO MONITOR select switches
- To select the audio output for the PHONES jack and the AUDIO MONITOR OUT connector.
- The Hi-Fi/NORM switch also switches the audio level meters between Hi-Fi and NORMAL.

  Hi-Fi: To monitor the Hi-Fi audio signals.
  NORM: To monitor the normal audio signals.
  AUD-1/L: To monitor the normal audio-1 or Hi-Fi left-channel signal.
  MIX: To monitor the AUD-1/L and AUD-2/R signals together.
  AUD-2/R: To monitor the normal audio-2 signal or Hi-Fi right-channel signal.

㉕ VIDEO INPUT select switch
- To select an input video signal for recording.

  Y/C358: To record the signal input to the Y/C358 connector.
  LINE: To record the signal input to the VIDEO IN LINE connector.
  BLACK: To record the internally-generated black burst signal on a blank tape in preparation for insert editing. If set to this position during menu setting, on-screen information is output from all output connectors, not only the MONITOR OUT connector.

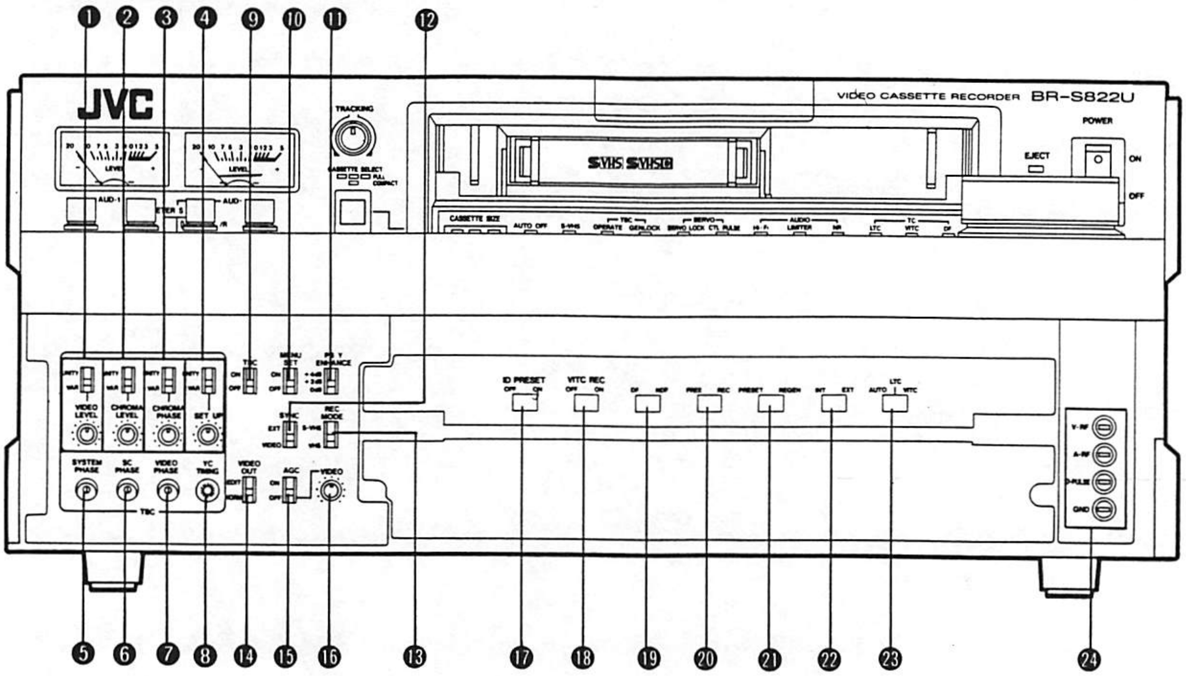
㉖ REMOTE select switch
- To select between remote and local control of the recorder.

  9-PIN: For remote control via the rear panel 9-pin connector.
  LOCAL: For direct control with the recorder's function buttons.
  REM-2: For remote control via the optional 45-pin or RS-232C interface.

㉗ COUNTER select switch
- To select the time counter display mode with the SA-R22U TC generator/reader installed. If this is not installed, CTL signals are displayed regardless of the switch setting.

  CTL: CTL signals are displayed on the time counter.
  TC: Time code signals are displayed on the time counter.
  UB: User bits are displayed on the time counter.

FRONT SUB-PANEL



TBC CONTROLS

The controls in this section function when the optional SA-T22U TBC (time base corrector) is installed.

1 VIDEO LEVEL UNITY/VARIABLE select switch/level control

UNITY: The output signal's video level is the same as the playback signal. Normally set to this position.

VARIABLE: Allows you to adjust the output signal's video level with the VIDEO LEVEL control. Adjustment is possible within ±3 dB.

2 CHROMA LEVEL UNITY/VARIABLE select switch/level control

UNITY: The output signal's chroma level is the same as the playback signal. Normally set to this position.

VARIABLE: Allows you to adjust the output signal's chroma level with the CHROMA LEVEL control. Adjustment is possible within ±3 dB.

3 CHROMA PHASE UNITY/VARIABLE select switch/level control

UNITY: The output signal's chroma phase is the same as the playback signal.

VARIABLE: Allows you to adjust the output signal's chroma phase with the CHROMA PHASE control. Adjustment is possible within ±30°.

4 SET UP VARIABLE/UNITY select switch/level control

UNITY: The output signal's setup level is the same as the playback signal.

VARIABLE: Allows you to adjust the output signal's setup level with the SET UP control. Adjustment is possible within ±15 IRE.

5 SYSTEM PHASE control

- Adjusts the output signal's horizontal phase with respect to that of the reference input signal. Adjustment is possible within a range of ±3 μsec.

6 SC PHASE

- Adjusts the output signal's subcarrier phase with respect to that of the reference input signal. Up to 15 rotations are possible with continuous variation over a range of ±180°.

7 VIDEO PHASE control

- Adjusts the output signal's video phase with respect to the playback signal's H sync. Up to 15 rotations are possible with continuous variation over a range of ±1.5 μsec.

8 YC TIMING control

- Adjusts the output signal's C signal delay time with reference to the Y signal. Adjustable within ±500 nsec.
- Normally set to "8".

⑨ **TBC ON/OFF switch.**
- Set to ON for TBC playback. (During TBC operation, the servo is locked to the reference signal supplied to the EXT REF connector even if the SYNC select switch is set to VIDEO.)
- Pressing any of the edit select buttons defeats TBC operation.
- Set to OFF to bypass TBC.

⑩ **MENU SET ON/OFF switch**
- SET to ON to activate the On-Screen Menu. The counter display will also switch to the Menu Set mode.
- Most basic system setup operations are performed using the Menu.

⑪ **PB Y ENHANCE switch**
- Enhances the luminance signal for a sharper playback picture.

+4 dB: Boosts luminance signal level by 4 dB at 2.5 MHz for maximum picture sharpness.
+2 dB: Boosts luminance signal level by 2 dB at 2.5 MHz for a sharper picture.
0 dB: No effect. The same result is obtained by setting the VIDEO OUT select switch ⑭ to EDIT.

⑫ **SYNC select switch**
EXT: The servo is synchronized with the external reference signal supplied to the EXT REF input.
VIDEO: The servo is synchronized with the input video signal.

⑬ **REC MODE select switch**
S-VHS: To record in the S-VHS mode. (Use S-VHS cassettes only)
VHS: To record in the VHS mode.

⑭ **VIDEO OUT select switch**
EDIT: Set to this position when using this VCR as a feeder or recorder in editing.
NORM: Normally set to this position.

⑮ **VIDEO AGC ON/OFF switch**
- Set to ON to activate the built-in VIDEO AGC circuit.
- Set to OFF to adjust the luminance video recording level manually.

⑯ **VIDEO control**
- Use to adjust video recording level, referring to the VIDEO/TRACKING meter. The center click-stop is the standard position. The VIDEO AGC switch must be OFF to use this control.

TIME CODE GENERATOR/READER SETTING SWITCHES (With SA-R22U TC generator/reader installed)

⑰ **ID PRESET ON/OFF switch**
ON: To record the ID code specifically preset for each VCR.
OFF: To use the user bits memory for standard procedures in the Preset mode.

⑱ **VITC REC ON/OFF switch**
ON: To record VITC time codes.
OFF: VITC time codes are not recorded.

NOTE:
This switch has no effect on LTC recording (enabled by setting menu item #206 to "01 – LTC").

⑲ **DF/NDF switch**
- This switch is effective only when the PRESET/REGEN switch is set to PRESET and the INT/EXT switch is set to INT.

DF: To record time data (TC or CTL) in the drop-frame mode.
NDF: To record time data (TC or CTL) in the non-drop-frame mode.

⑳ **FREE/REC switch**
- This switch is effective only when the PRESET/REGEN switch is set to PRESET and the INT/EXT switch is set to INT.

FREE: The time code runs in real time, regardless of the video recorder's operating mode.
REC: The time code runs only during recording.

㉑ **PRESET/REGEN switch**
PRESET: To use the internal TC generator in the Preset mode (with the INT/EXT switch set to INT), or to use an external TC generator via the TIME CODE IN/OUT connectors (with the INT/EXT switch set to EXT).
REGEN: To use the internal TC generator in sync with either the playback time codes (with the INT/EXT switch set to INT), or externally input time codes (with the INT/EXT switch set to EXT).

㉒ **INT/EXT switch**
INT: To use the internal TC generator/reader.
EXT: To use an externally-connected LTC generator/reader.

㉓ **AUTO/LTC/VITC switch**
- To select the TC reader mode. Select the mode according to the type of reference time code with which the internal TC generator is synchronized in the Regen mode.

AUTO: For tapes with matching VITC and LTC data. Counts time codes in VITC at tape speeds lower than normal, and in LTC at speeds higher than normal. Missing sections are interpolated with CTL counts.
LTC: For LTC-only tapes or when editing with LTC data. Counts time codes in CTL at tape speeds lower than normal, and in LTC at speeds higher than normal. Missing sections are interpolated with CTL counts.
VITC: For VITC-only tapes or when editing with VITC data. Counts time codes in VITC at tape speeds lower than normal, and in CTL at speeds higher than normal. Missing sections are interpolated with CTL counts.

㉔ **Test points**
V-RF test point
- Outputs the video head FM signal during playback.
- Can be used for detection of clogged or worn heads.

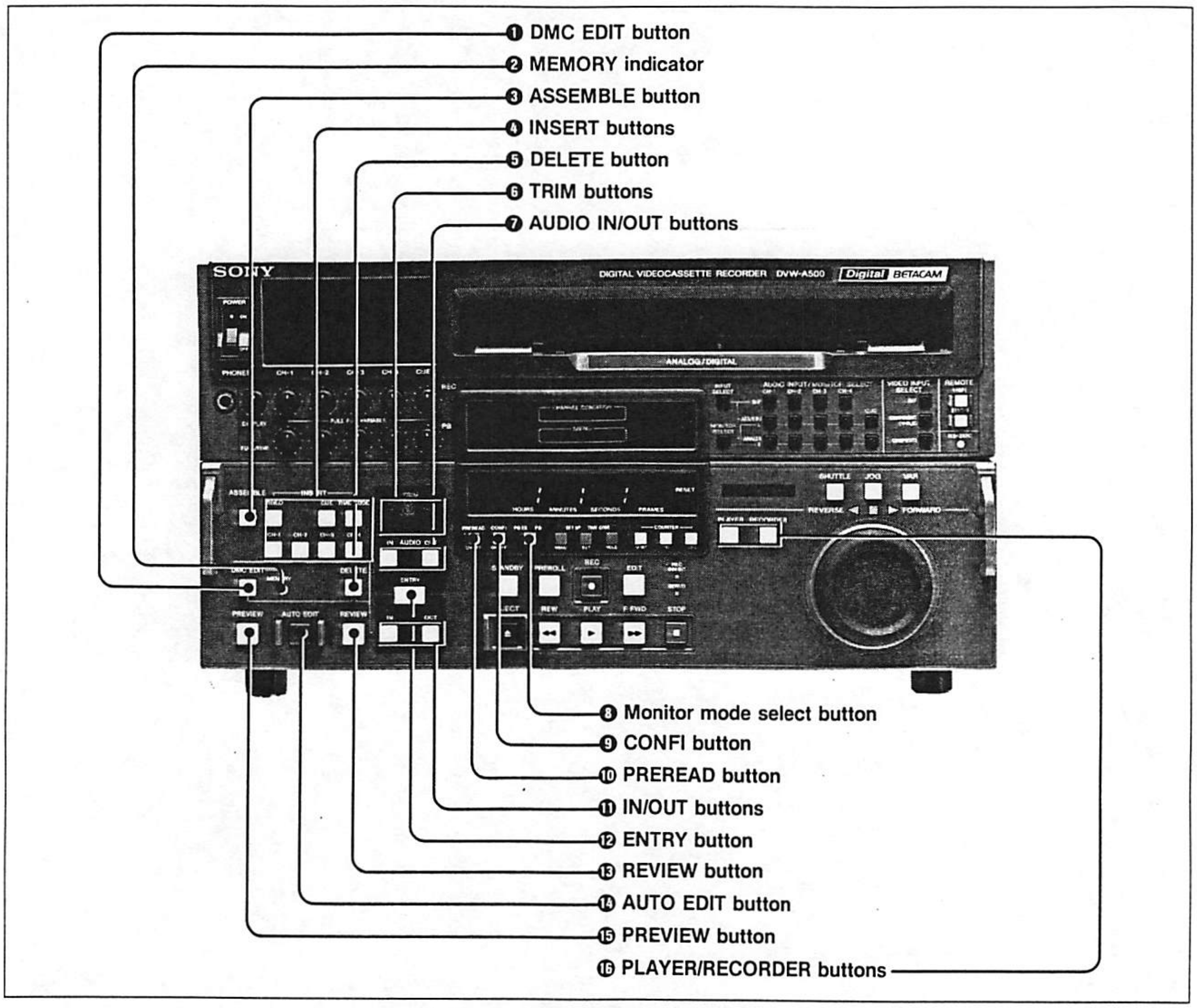
A-RF test point
- Outputs the Hi-Fi audio FM signal during playback.
- Can be used for detection of clogged or worn heads.

D-PULSE pin
- Connect to the external trigger terminal of an oscilloscope.

GND
- Connect to the ground terminal of an oscilloscope.

Chapter 2

2-1-2. Lower Control Panel (Editing Section)



Lower control panel (editing section)

❶ DMC EDIT button
Performs DMC playback or DMC editing at speeds between −1 and +3 times normal speed as stored in memory.

❷ MEMORY indicator
Flashes to show that playback speeds are being stored in memory while performing DMC playback or DMC editing using the DMC EDIT button. Lights steadly once the speeds have been memorized.

❸ ASSEMBLE button
Selects assemble edit mode.
This button lights when pressed and goes off when pressed again.

❹ INSERT buttons
Select the input signal channel (VIDEO, AUDIO CH-1, CH-2, CH-3, CH-4, CUE or TIME CODE) for insert edit.
These buttons light when pressed and go off when pressed again.

❺ DELETE button
Deletes an edit point or exits DMC mode.
Pressing this button together with a lit IN, OUT, AUDIO IN or AUDIO OUT button deletes the edit point corresponding to that button. The button pressed together with this button will go off or flash. If it flashes, another edit point should be set.

❻ TRIM buttons
Changes an edit point by one frame.
Press either of the TRIM +/− buttons together with the IN, OUT, AUDIO IN or AUDIO OUT button. Pressing the + button advances the edit point by one frame, while pressing the − button moves it back one frame.
Pressing the + or − button together with the PLAY button changes the playback speed by +8% or −8% (capstan override function).

❼ AUDIO IN/OUT buttons
Set the audio IN and OUT edit points separately from the video edit points.
Press one of these buttons together with the ENTRY button to set an audio IN/OUT point.
If you press one of these buttons after setting an audio edit point, the corresponding time data will be shown in the time counter display.

❽ Monitor mode select button
Selects the signal output while the VTR is in fast forward, rewind, stop or standby mode. Press this button to turn on the desired indicator.
PB/EE: Input signal
PB: Playback signal
When playing back an analog Betacam tape, the monitor mode is automatically set to PB regardless of your selection.

❾ CONFI (confidence) button
Enables you to monitor the video and audio currently being recorded by simultaneously playing back them with the dedicated confidence heads.

❿ PREREAD button
Performs pre-read (read before write) in insert editing mode.

⓫ IN/OUT buttons
Set an IN/OUT edit point.
Press one of these buttons together with the ENTRY button.
If you press one of these buttons after setting an edit point, the corresponding time data will be shown in the time counter display.

⓬ ENTRY button
Sets an edit point.
Press this button together with the IN, OUT, AUDIO IN or AUDIO OUT button. The IN/OUT buttons are for setting video IN/OUT edit points and the AUDIO IN/OUT buttons are for audio IN/OUT edit points. The button pressed together with this button lights.

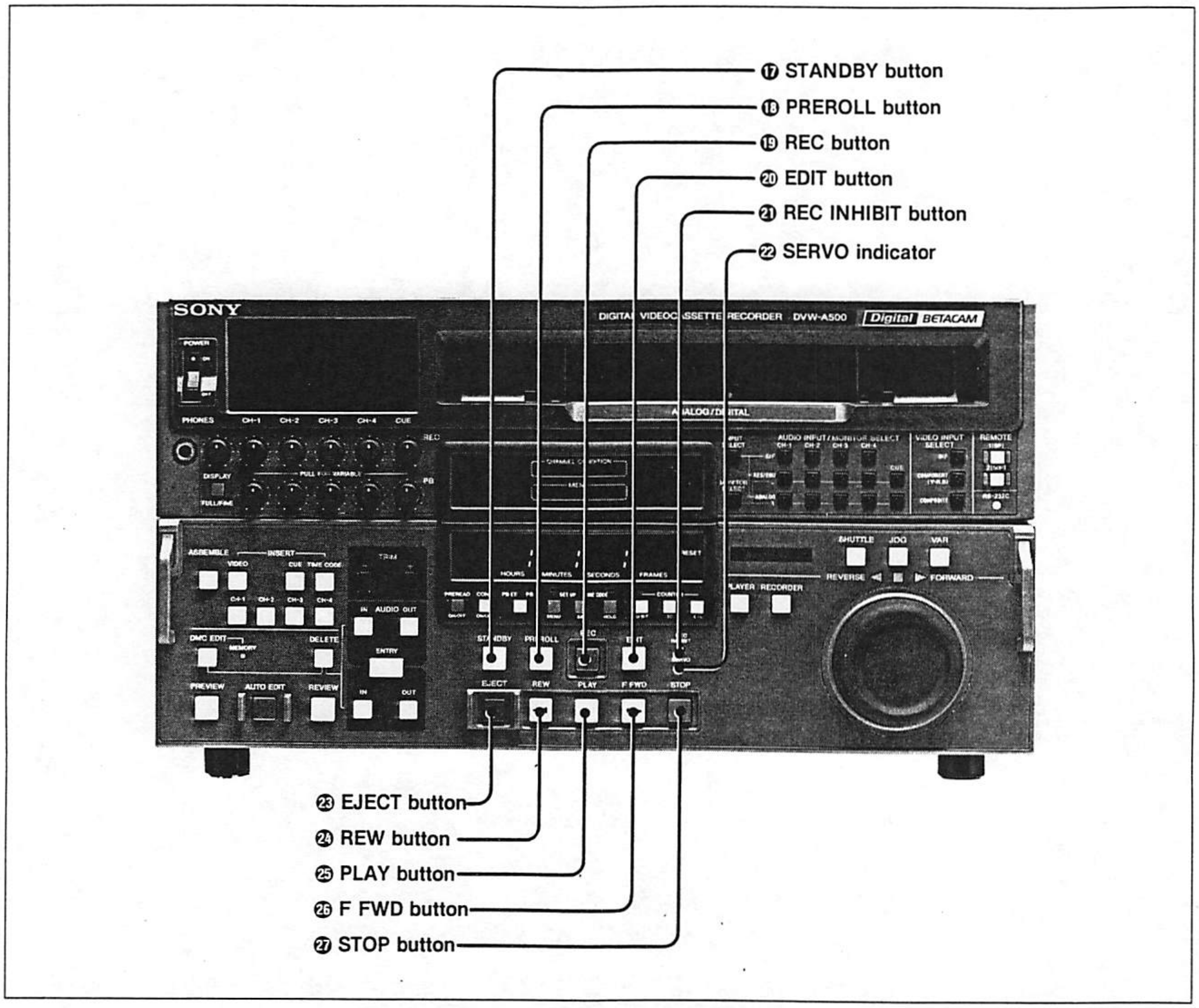
⓭ REVIEW button
Reviews a section of the edit on the recorder monitor.

⓮ AUTO EDIT button
Starts automatic editing.
If an IN point is not set when you press this button, the current tape address is set as the IN point before automatic editing starts.

⓯ PREVIEW button
Lets you view the results of an edit on a monitor connected to the recorder without actually recording the edit to tape.
If an IN point is not set when you press this button, the current tape address is set as the IN point before the preview starts.

⓰ PLAYER/RECORDER buttons
Select which of the VTRs is to be controlled by the editing buttons and tape transport buttons of this VTR when this VTR is configured as a recorder with an external player VTR connected to the REMOTE 1-IN (9P)/OUT (9P) connectors.
PLAYER button: When this button is pressed, it lights to show that the editing and tape transport buttons of this VTR control the external player VTR.
RECORDER button: When this button is pressed, it lights to show that the editing and tape transport buttons control the recorder VTR (this VTR).
These buttons have no effect when using the VTR alone.

2-1-3. Lower Control Panel (Tape Operation Section)



Lower control panel (tape operation section)

⑰ STANDBY button

Pressing this button in other than standby mode makes it light and places the VTR in standby mode. Since the head-drum rotates in standby mode, the tape can thus start sooner than in non-standby mode.

Pressing this button while in standby mode turns it off, the VTR exits standby mode, the tape tension is released, and the drum stops rotating. If more than eight minutes (factory preset) elapse while the VTR is set to standby mode, the VTR automatically exits standby mode to prevent tape and head clogging.

⑱ PREROLL button

Pressing this button positions the tape to the preroll point (factory preset to five seconds prior to the IN point).

Use this button to cue up a tape for broadcast or manual editing.

Pressing this button together with the IN, OUT, AUDIO IN or AUDIO OUT button cues up the tape to the edit point corresponding to the button.

For details of changing the preroll time, see "3-3. Setup Menu" on page 3-8.

⑲ REC (record) button
Pressing this button together with the PLAY button starts recording.
Holding down the REC button during playback, search, fast forward or rewind allows you to monitor the E-E video and audio.
Pressing the REC button while the VTR is in stop mode allows you to monitor the E-E video and audio. Pressing the STOP button while monitoring E-E signals returns you to the video and audio monitored before you pressed the REC button.

⑳ EDIT button
Pressing this button together with the PLAY button starts manual editing.
Pressing the EDIT button while the VTR is in stop mode allows you to monitor E-E signals selected with the ASSEMBLE or INSERT buttons.
Pressing the STOP button while monitoring the input signal release the edit E-E mode.
Holding down the EDIT button during playback, search, fast forward or rewind allows you to monitor E-E signals.

㉑ REC (record) INHIBIT indicator
Lights or goes off according to the setting of the REC INHIBIT switch on the sub control panel and the status of the record inhibit plug of the cassette, as shown below.

Status of REC INHIBIT indicator

REC INHIBIT switch	Record inhibit plug	REC INHIBIT indicator
ON	Pushed down/ Not pushed down	Lit
OFF	Pushed down	Lit[a]
	Not pushed down	Unlit

a) You can change the initial setting such that the indicator flashes here.

For details, refer to "1-10. Setup Menu" in the Installation Manual.

Recording, editing and the selection of assemble or insert mode are possible only when this indicator is off.

㉒ SERVO indicator
Lights when the drum servo and capstan servo lock.

㉓ EJECT button
Ejects the cassette. Resets the display when CTL codes appear in the time counter display.

㉔ REW (rewind) button
Rewinds the tape.

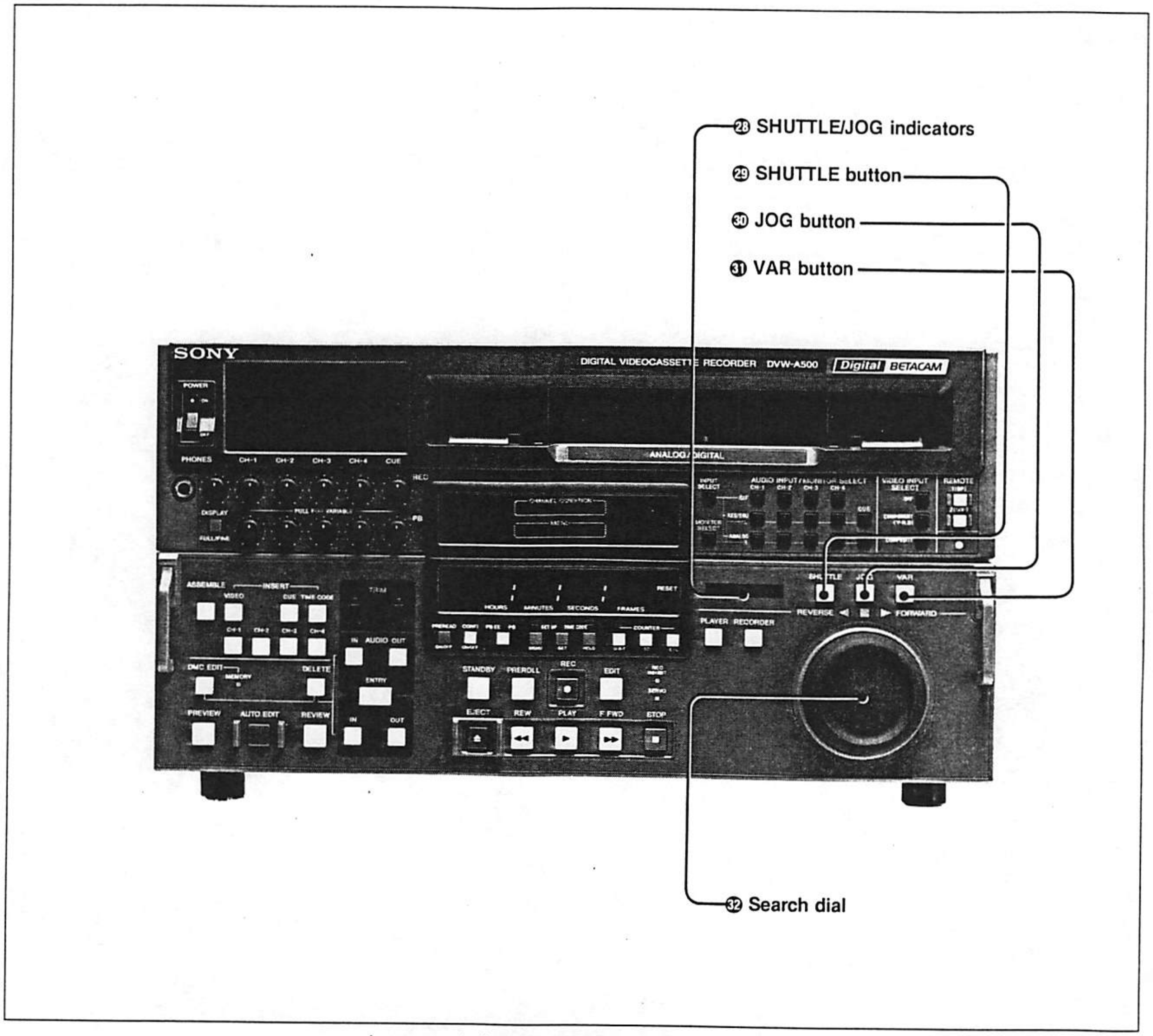
㉕ PLAY button
Starts playback.
Pressing this button together with the REC button or EDIT button starts recording or manual editing. Pressing this button during recording or manual editing places the VTR in playback mode.

㉖ F FWD (fast forward) button
Fast forwards the tape.

㉗ STOP button
Stops the tape (stop mode).
When the monitor mode is set to PB, pressing this button outputs a still picture. In stop mode, the tape is wound around the head-drum, but the head-drum continues to rotate. The VTR enters stop mode when a cassette is loaded.
This button flashes when the input video signal and the external reference signal are out of phase or when each signal is not supplied in the VIDEO INPUT and REF mode respectively. This indication can be switched off by setup menu item 105.

For details, see "1-10. Setup Menu" in the Installation Manual.

2-1-4. Lower Control Panel (Search Operation Section)



Lower control panel (search operation section)

㉘ **SHUTTLE/JOG indicators**
One of these indicators lights to show the current search mode.

㉙ **SHUTTLE button**
Selects shuttle mode to play back the tape at 0 to ±50 times normal speed when playing back a Digital Betacam tape, and ±35 (DVW-A500) or ±42 (DVW-A500P) times normal speed when playing back an analog Betacam tape. The SHUTTLE indicator lights. The dial is indented at the positions corresponding to 0, −10 and +10 times normal speed. The playback speed corresponds to the angle of rotation of the dial.

㉚ **JOG button**
Selects jog mode to play back the tape at 0 to ±1 or ±3 times normal speed (selectable with the menu setup). The JOG indicator lights. The dial is not indented. The playback speed corresponds to the rotational speed of the dial.

㉛ **VAR button**
Selects variable speed playback mode with noiseless picture at any of 54 speeds within the range −1 to +3 times normal speed. The VAR indicator lights. The dial is indented at the positions of still and normal playback speed.

㉜ **Search dial**
Searches for edit points.
Rotate the dial clockwise for forward playback (▶ indicator lit) or counterclockwise for reverse playback (◀ indicator lit). While the VTR is in stop mode, the ■ indicator lights.
When you press the dial, the VTR toggles between shuttle and jog modes, and either the SHUTTLE or JOG indicator lights to indicate the current mode.

Shuttle mode: The playback speed corresponds to the angle of rotation of the dial (0 to about ±50 times normal speed when playing back a Digital Betacam tape, and ±35 (DVW-A500) or ±42 (DVW-A500P) times normal speed when playing back an analog Betacam tape.).

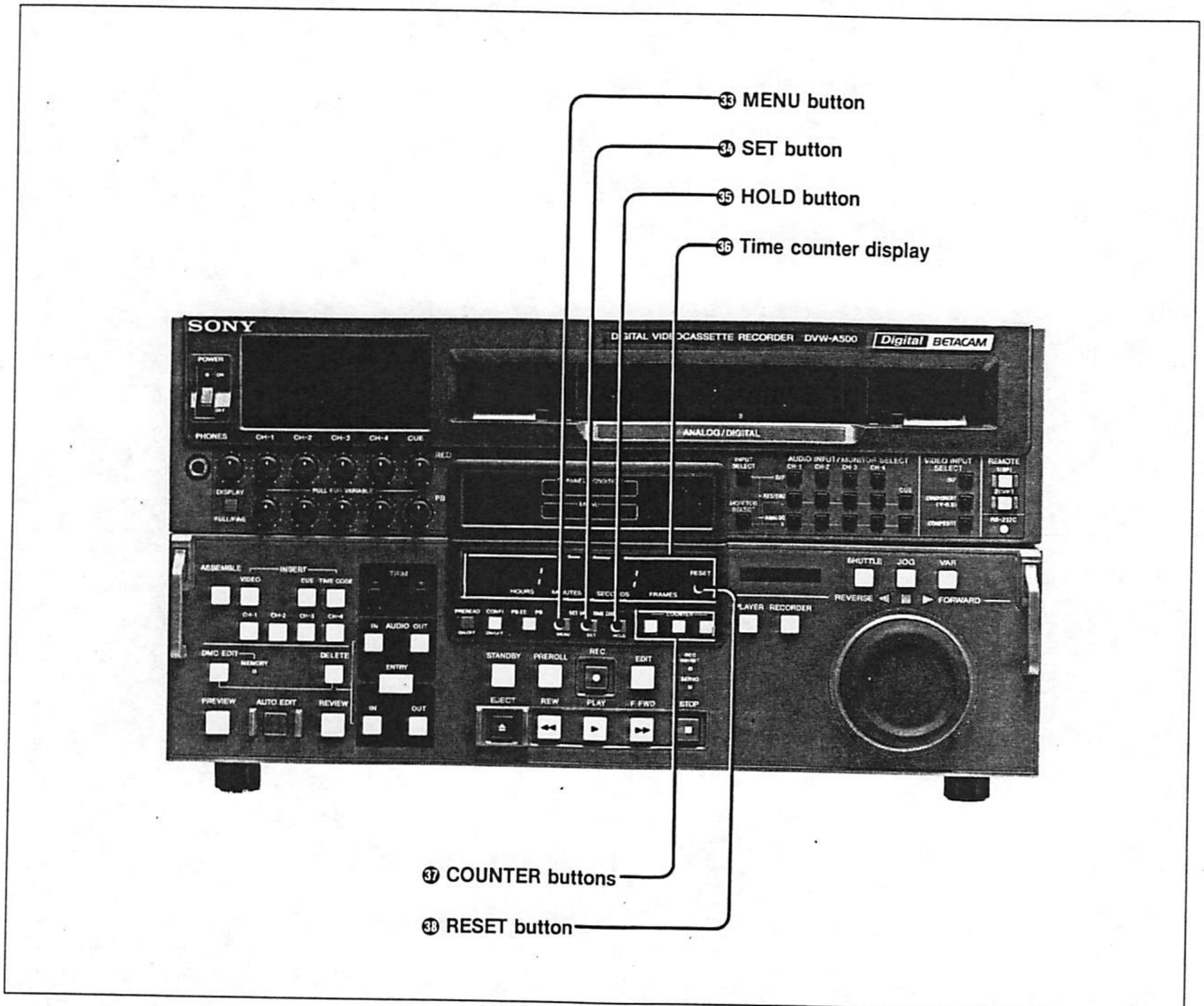
Jog mode: The dial is not indented. The playback speed corresponds to the rotational speed of the dial (0 to ±1 or ±3 times normal speed).

VAR: Playback speed is −1 times normal speed when the dial is rotated fully counterclockwise, and +3 times normal speed when rotated fully clockwise.

Capstan override mode: Rotating the dial while holding down the PLAY button changes the playback speed by up to ±15%.

After turning on the power, always center the search dial (where the indicator lights).

2-1-5. Lower Control Panel (Time Code Section)



Lower control panel (time code section)

㉝ MENU button
When you press this button, the indicator will light and a menu will be displayed. If you press this button again, the indicator will go out, and any changes will not be saved.

㉞ SET button
Press this button to register the value shown in the time counter display with the time code generator, after setting the value using the search dial.
If you press this button after changing one or more items in the menu, those changes will be saved.

㉟ HOLD button
Press this button to temporarily halt the progress of the data in the time counter display. The indicator lights and the data in the time counter display freezes. Press the button again to resume display of the data presently being read. You should press this button first before setting the time code or user's bits.

㉟ Time counter display

Displays the following time data according to the setting of the time counter display switch.

CTL: Tape running time in hours, minutes, seconds and frames, as determined by counting the CTL signal currently being recorded onto the tape or read from the tape currently being played back.

TC: Time code currently being recorded onto or read from the tape by the built-in time code reader. Either the LTC or VITC time code is displayed, according to the setting of the TC selector on the system set-up panel.

U-BIT: User's bits included in the time code currently being recorded onto or read from the tape. User's bits included in LTC or VITC time codes are displayed, according to the setting of the time code selector on the system set-up panel.
One of these indicators lights to show the current search mode.

For the meanings of the error codes ("Error-XX") displayed in the time counter display, refer to the Maintenance Manual Part 1.

㊲ COUNTER buttons

Press one of the buttons corresponding to the type of data to be displayed in the time counter display.

COUNTER buttons, data displayed and editing tape address

The COUNTER buttons pressed	Data displayed	Editing tape address
U-BIT	User's bits	Time code
TC	Time code	Time code
CTL	CTL	CTL

When the REMOTE 1 (9P) button is pressed, the time data is displayed and the editing tape address is determined according to the equipment connected to the REMOTE 1 connector, regardless of the setting of these buttons.

㊳ RESET button

Resets the time counter as follows, according to the setting of the time counter display switch:

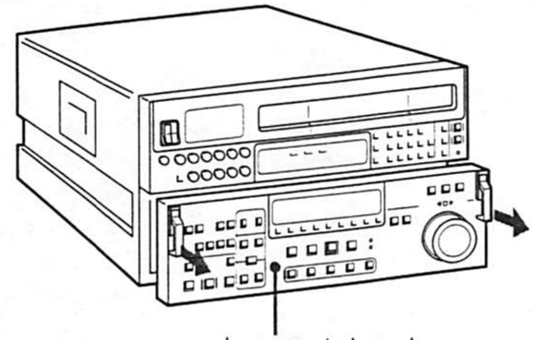
CTL: Sets the CTL display to 0:00:00:00. All currently set edit points are erased.

TC or U-BIT: Resets the time code generator and sets the time code to 00:00:00:00 (when set to TC) and user's bits to 00 00 00 00 (when set to U-BIT). Currently set edit points are not affected.

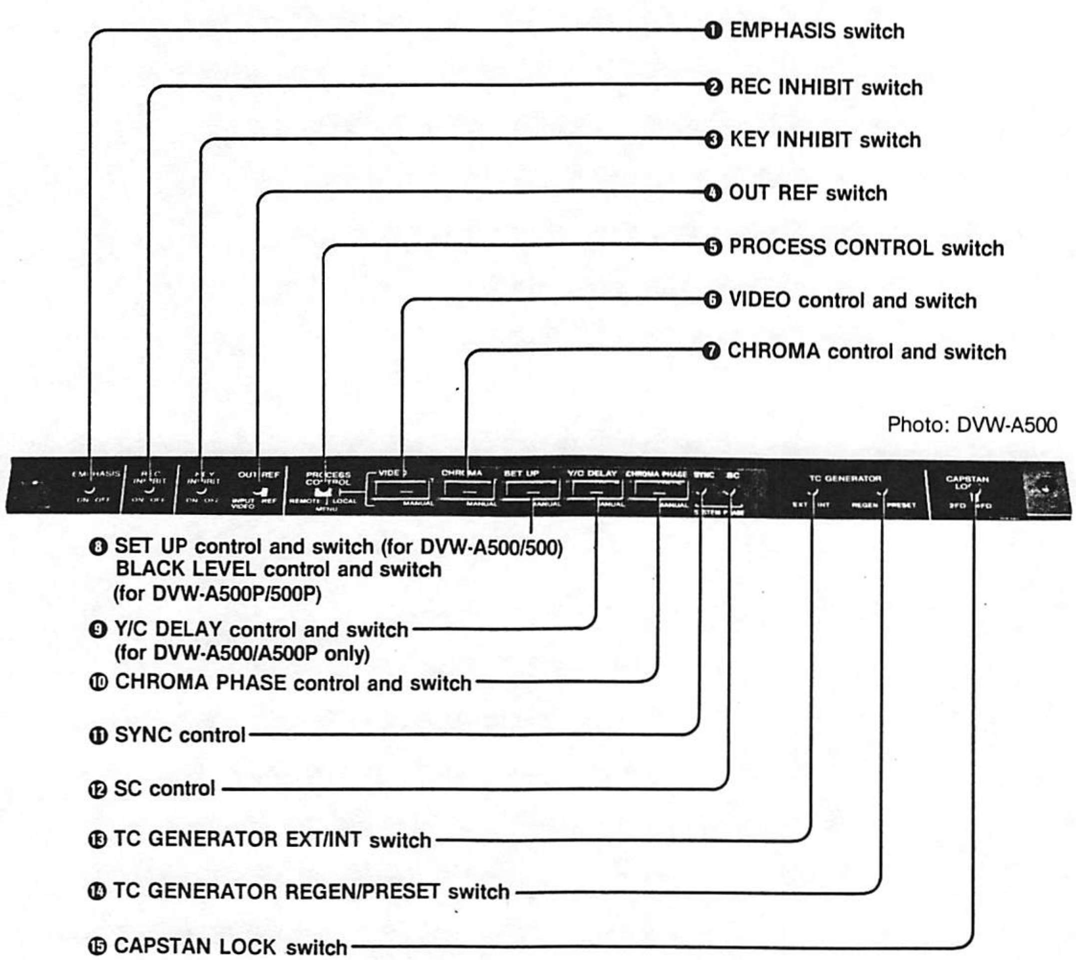
2-2. Sub Control Panel

105

Pull out the lower control panel to access the sub control panel.



How to access the sub control panel



Sub control panel

❶ **EMPHASIS switch**
Set this switch to ON in case of emphasizing digital audio signals converted from the BETACAM/BETACAM SP playback signals.

❷ **REC INHIBIT switch**
Setting this switch to ON turns on the REC INHIBIT indicator on the lower control panel and inhibits recording, editing and the selection of assemble or insert edit mode.

❸ **KEY INHIBIT switch**
Setting this switch to ON turns on the KEY INHIBIT indicator of the upper control panel and inhibits all or some key inputs of the REMOTE, VIDEO INPUT SELECT, AUDIO INPUT/MONITOR SELECT buttons, and the buttons in the editing section.
The inhibited key inputs can be changed with the setup menu.
For details, refer to "1-10. Setup Menu" in the Installation Manual.

❹ **OUT REF switch**
Selects the reference signal with which the output signals are synchronized.
REF: Selects the external reference signal to be input to the REF. VIDEO connector as the reference signal for playback and digital audio recording. The input video and digital audio signal should be synchronized with the external reference signal when recording.
If they are out of phase or no reference signal is supplied, the STOP button flashes to warn that.
INPUT VIDEO: Selects the input video signal to be used as the reference signal for both playback and recording. To select the input video signal, press one of the VIDEO INPUT SELECT buttons.

❺ **PROCESS CONTROL switch**
Selects the equipment that controls the digital video processor.
Switching this switch during playback may cause momentary muting in the playback sound.
LOCAL: The video processor is set according to the settings of the controls and switches on the sub control panel of the VTR.
MENU: The video processor is set according to the setting of setup menu items 715 to 721.
For details, refer to "1-10. Setup Menu" in the Installation manual.
REMOTE: The optional BVR-50/50P remote control unit controls the video processor.

❻ **VIDEO control and switch**
Adjusts the output level of the video signal. Set the VIDEO switch as follows.
MANUAL: To manually adjust the output level of video signals by using the VIDEO control. The adjustment range is ±3 dB, or −∞ to +3 dB according to the setting of setup menu item 714. Note that only the ±3 dB range is available when controlled from the BVR-50/50P.
PRESET: To set the output level of video signal to the preset standard level. The setting of the VIDEO control is ignored.

❼ **CHROMA control and switch**
Adjusts the chrominance signal output level. Set the CHROMA switch as follows.
MANUAL: To manually adjust the chrominance signal output level by using the CHROMA control. The adjustment range is ±3 dB, or −∞ to +3 dB according to the setting of setup menu item 714.
Note that only the ±3 dB range is available when controlled from the BVR-50/50P.
PRESET: To set the chrominance signal output level to the preset standard level. The setting of the CHROMA control is ignored.

❽ **SET UP control and switch (for DVW-A500/500)**
Adjusts the setup level (black level). Set the SET UP switch as follows.
MANUAL: To manually adjust thean setup level using the SET UP control. The adjustment range is ±30 IRE.
PRESET: When the setup level is not to be adjusted. The setting of the SET UP control is ignored.

❽ **BLACK LEVEL control and switch (for DVW-A500P/500P)**
Adjusts the black level. Set the BLACK LEVEL switch as follows.
MANUAL: To manually adjust the black level using the BLACK LEVEL control. The adjustment range is ±210 mV.
PRESET: When the black level is not to be adjusted. The setting of the BLACK LEVEL control is ignored.

❾ **Y/C DELAY control and switch (for DVW-A500/A500P only)**
Adjusts the Y/C delay of analog Betacam or Betacam SP playback. Set the Y/C delay switch as follows.
MANUAL: To manually adjust the Y/C delay using the Y/C DELAY control. The adjustment range is ±100 ns.
PRESET: When the Y/C delay is not to be adjusted. The setting of the Y/C DELAY control is ignored.

⑩ CHROMA PHASE control and switch
Adjusts the hue (burst and chroma relative phase). Set the CHROMA PHASE switch as follows.
MANUAL: To adjust the hue within ±30 degrees with the control.
PRESET: When the hue is not to be adjusted, regardless of the control setting.

⑪ SYNC control
Adjusts the output sync phase by up to ±15 μs with respect to the reference signal being input to this unit in E-E and playback mode. This control is inhibited while recording. Note that momentary muting on playback sound may occur when adjusting the sync phase during playback.

⑫ SC (subcarrier) control
Adjust the output sync and subcarrier phase within ±200 ns with respect to the reference signal being input to this unit. Use this control if you need to precisely adjust the unit's output in subcarrier phase order with the reference signal for editing composite signals. The output SCH (subcarrier to sync) phase is not changed with this control.

⑬ TC GENERATOR EXT/INT (time code external/internal) switch
Determines whether an external or internal time code is used.
EXT: The time code being input to the TIME CODE IN connector is used.
INT: The time code generated by the internal time code generator is used.
This switch is factory set to INT.

⑭ TC GENERATOR REGEN (regeneration) /PRESET switch
Selects the time code with which the internal time code generator synchronizes.
REGEN: Time code read by the time code reader.
PRESET: Time code set with the VTR.

⑮ CAPSTAN LOCK switch
When editing or playing back a tape on which decoded component signals are recorded (a tape on which composite signals were recorded with a Digital Betacam or a Betacam and Betacam SP VTR), set this switch according to your editing or playback requirements.

For DVW-A500/500
2FD: Capstan servo locks in units of 2 fields. Since color framing lock is inhibited, there is no phase shift (H-shift) of the output video signal. Set to 2FD for component signal editing/playback. However, optimum composite signal frequency response can be obtained by shifting the video phase (H-shift) while refering to the decoded SC phase or the color frame ID, which is enabled by using the setup menu item 712 for composite editing with quick servo lock.
4FD: Capstan servo locks in units of 4-field color frame. Optimum composite frequency response is kept and there is no picture shift at any edit point or stop/start point during playback.
Set to 4FD for composite editing/playback or A-B roll editing that requires stable and continuous video phase.

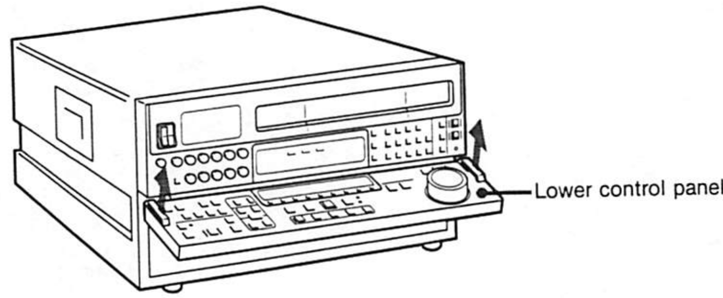
For DVW-A500P/500P
2FD: Capstan servo locks in units of 2 fields. Since color framing lock is inhibited, there is no phase shift (H-shift) of the output video signal. Set to 2FD for component editing/playback. However, optimum composite frequency response can be obtained by shifting the video phase (picture shift) while referring to the decoded SC phase or the color frame ID, which is enabled by using the setup menu item 712 for composite editing with quick servo lock.
4FD: Capstan servo locks in units of 4 fields. Optimum composite frequency response can be obtained by shifting the video phase (H-shift) while referring the decoded SC phase or the color frame ID. Set to 4FD for composite signal editing with quick servo lock.
8FD: Capstan servo locks in units of 8-field color frame. Optimum composite signal frequency response is kept and there is no picture shift or V-shift at any edit point or stop/start point during playback.
Set to 8FD for composite editing/playback or A-B roll editing that requires stable and continuous video phase.

Note
In playback of Digital Betacam tape on which component source is recorded, picture shift control is inhibited in any mode.
As for composite source recording with the BKDW-505/506, the video phase is automatically adjusted so that SCH phase becomes zero if the phase is shifted.

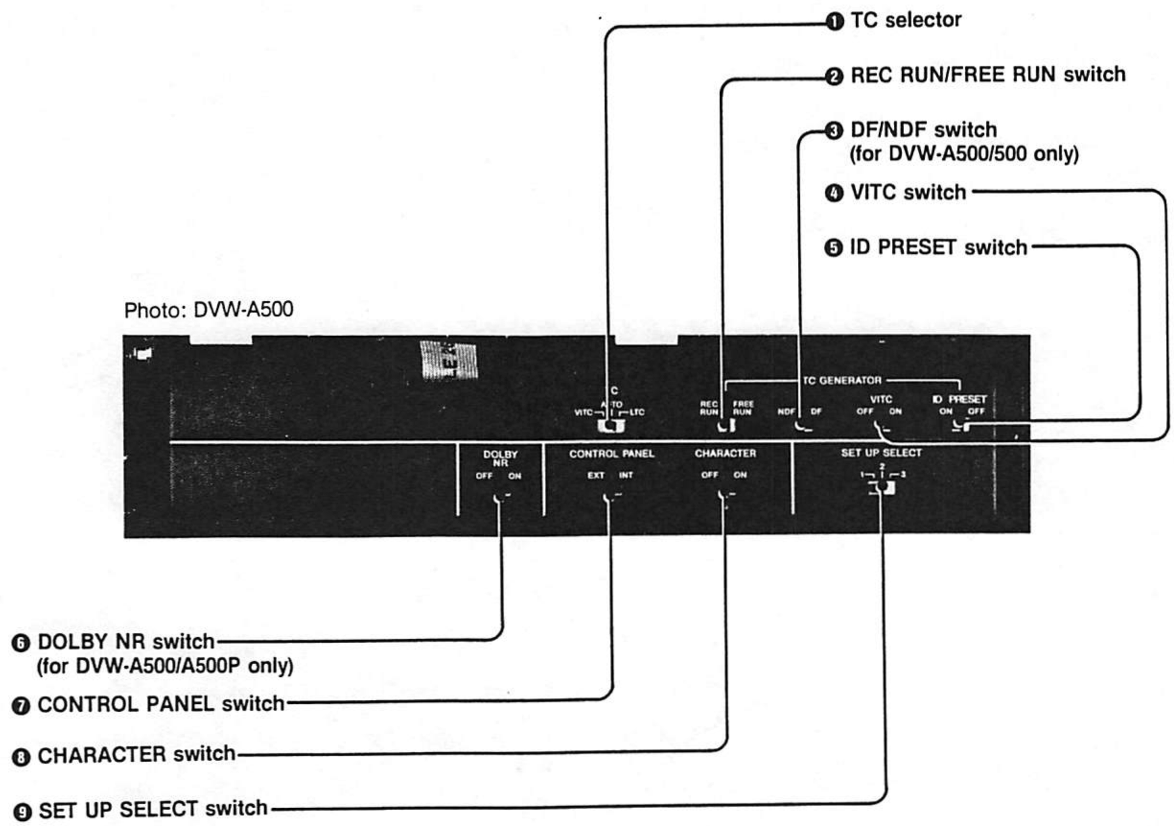
2-3. System Set-Up Panel

108

Lift the lower control panel up into its horizontal position to access the system set-up panel.



How to access the system set-up panel



System set-up panel

❶ TC (time code) selector
Determines whether the time code displayed in the time counter display on the lower control panel is LTC or VITC. When this switch is set to AUTO, the time counter display shows LTC when the tape speed exceeds ±1/2 times normal speed, but VITC when the speed is within ±1/2 times normal speed.
This switch is factory set to LTC.

❷ REC RUN/FREE RUN switch
Selects the operation mode of the time code generator.
REC RUN: The time code advances only during recording. When you set the switch to this position, set the TC GENERATOR EXT/INT switch on the sub control panel to INT and the TC GENERATOR REGEN/PRESET switch on the sub control panel to PRESET.
FREE RUN: Regardless of the operation mode of the VTR, the time code advances as long as the VTR is powered on.
This switch is factory set to FREE RUN.

❸ DF/NDF (drop frame/non drop frame) switch (for DVW-A500/500 only)
Selects whether the time code generator and CTL counter advance in drop frame mode or non-drop frame mode.
NDF: Non-drop-frame mode.
DF: Drop frame mode.
This switch is factory set to DF.
If the TC GENERATOR REGEN/PRESET switch is set to REGEN, this switch has no effect since the time code generator synchronizes with the external or playback time code.

❹ VITC (Vertical Interval Time Code) switch
Selects whether VITC signals are recorded.
OFF: VITC signals are not recorded.
Note that VITC signals in the input video signals are automatically recorded.
ON: The VITC generated by the built-in time code generator is recorded.
This switch is factory set to ON.

For details of the VITC insertion line, refer to "1-10. Setup Menu" in the Installation Manual.

❺ ID PRESET switch
Sets the user's bits generated by the time code generator to the ID code value preset in the set up menu.
This switch is factory set to OFF.

For details of setting and storing the ID code, refer to "1-10. Setup Menu" in the Installation Manual.

❻ DOLBY NR switch (for DVW-A500/A500P only)
Turns on and off the Dolby C noise reduction system when using Betacam format oxide tape.
ON: Dolby NR system is used for Betacam playback.
OFF: Dolby NR system is disabled for Betacam playback.
This switch is factory set to OFF.

1)Dolby NR
Dolby noise reduction manufactured under license from Dolby Laboratories Licensing Corporation.
"DOLBY" and the double-D symbol □□ are trademarks of Dolby Laboratories Licensing Corporation.

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❼ CONTROL PANEL switch
Selects which control panel controls this VTR.
INT: The control panel on the front panel controls this VTR.
EXT: An external control panel, connected through the CONTROL PANEL connector, controls this VTR.
This switch is factory set to INT.

❽ CHARACTER switch
Selects whether character signals such as time codes are superimposed on the video signals output from the SERIAL V/A OUTPUT 4 (SUPER) and COMPOSITE VIDEO OUTPUT 3 (SUPER) connectors.
ON: Superimposed.
OFF: Not superimposed.
This switch is factory set to ON.

❾ SET UP SELECT switch
Selects the memory bank used to store set up menu settings for VTR operation. The VTR will be set according to the settings in the selected memory bank.
Pressing the MENU button displays the setup menu for the selected memory bank in the time counter display and/or on the monitor screen.

For details of the setup menu, refer to "3-3. Setup Menu" on page 3-8 and "1-10. Setup Menu" in the Installation Manual.

NTSC Video

The National Television Standards Committee (NTSC) developed a standard so that all TV signals in the United States would decode the same way on television sets. Otherwise, to watch different broadcasters like ABC or NBC, you would need a different TV set to decode the various TV signals. Standardization allowed the growth of the industry and a consumer investment in television. Any channel broadcast could be picked up as an NTSC standard video signal on a TV set and turned back into light information.

The specifications

Originally, NTSC was a black-and-white luminance channel. It played 30 frames every second, or 60 fields per second. It had 525 lines per frame, or 262.5 lines per field. The voltage specifications were: 1 volt from the lowest sync point to the highest (brightest) part of the picture information. *Blanking* (when the electron gun shuts off during retraces to the start of new lines and fields) was at about the middle of this 1 volt range. Black was represented as just slightly stronger voltage than blanking. This makes sense: the difference between the gun being off (blanked) and almost off (black) is quite small.

The new specifications

The new color NTSC standard had to make some slight adjustments to compensate for the new color information. These compensations could not be too great, however, or the NTSC signal would no longer play back on black-and-white TVs.

To prevent interference with the 3.58 MHz color subcarrier and to avoid crosstalk between the audio carrier frequencies, the frame rate was slightly reduced to 29.97 frames per second or 59.94 fields per second. All of the luminance, sync, and blanking information stayed the same.

Also, immediately following synchronization pulses, a 3.58 MHz burst of the color subcarrier is added to the video signal. This gives the receiving television a color subcarrier reference before every line of video. The TV decoding circuits compare the *phase* relationship of the color information in the picture to the color burst at the beginning of each line. Different phase relationships indicate different hues. The amplitude of the subcarrier in the picture information represents the saturation of the particular hue.

Basic NTSC color video specifications

When measuring video signals on a waveform monitor, these are the basic video signal specifications to check:

Synchronization pulses are at -40 IRE
Blanking level is at 0 IRE
Black level in the picture is 7.5 IRE
White level in the picture is 100 IRE

The vectorscope specifications are:

Color burst is at the 0° mark
All of the color *vectors* fall into their corresponding baskets or boxes on the waveform monitor graticule

What NTSC is and what it isn't
NTSC video is a composite video signal, meaning a luminance channel with a color subcarrier. This leads to crosstalk between the two signals. Sometimes luminance information can be mistaken for chrominance information, and the two can interfere with each other in a number of ways. It is considered a *reduced bandwidth* system because there is a 100% signal representing what was once three 100% signals (R, G, B).

NTSC video is not component video. It is only one channel.

NTSC video is an encoded signal derived from three color components (R, G, B) or a luminance channel and two color difference channels (Y, R-Y, B-Y).

"froze" new station construction for a time. But not even this could stop the boom. Television receivers continued to sell and sets in use increased at a phenomenal rate:

YEAR	SETS IN USE
1946	6,000
1947	142,000
1948	977,000
1949	3,660,000
1950	9,732,000
1951	15,637,000
1952	21,782,000

(Source: *Television Factbook*, Television Digest Inc., 1977)

The Debate Over Color

All of the receivers sold until this time were, of course, monochrome—black-and-white. But meantime the push for color was starting. In 1939 CBS had shown a color TV system in which a wheel with three color segments (filters) rotated at high speed in front of the camera lens. A similar wheel was used in the receiver. This system produced three picture fields (red, blue, and green) in rapid succession (hence the name "field sequential" system). Persistence of vision in the eye caused the viewer to see a single full-color picture. After the war CBS resumed work on this system and made a number of impressive demonstrations.

In 1949 the FCC started a series of hearings on proposed color TV standards. The commission viewed demonstrations of the CBS field sequential system, which worked very well, and of a new RCA all-electronic color system, which did not work quite so well. The RCA system used three separate pickup tubes in the camera and three kinescopes (one for each color) in the receiver. These required exact—and hard to maintain—alignment. However, this system had the big advantage of being "compatible" (i.e., a monochrome receiver tuned to the color transmission produced a good monochrome picture).

Announcing the

National Broadcasting Company, Inc.

National radio broadcasting with better programs permanently assured by this important action of the *Radio Corporation of America* in the interest of the listening public

The Radio Corporation of America is the largest distributor of radio receiving sets in the world. It handles the entire output in this field of the Westinghouse and General Electric factories.

It does not say this boastfully. It does not say it with apology. It says it for the purpose of making clear the fact that it is more largely interested, more selfishly interested, if you please, in the best possible broadcasting in the United States than anyone else.

Radio for 26,000,000 Homes

The market for receiving sets in the future will be determined largely by the quantity and quality of the programs broadcast.

We say quantity because they must be diversified enough so that some of them will appeal to all possible listeners.

We say quality because each program must be the best of its kind. If that ideal were to be reached, no home in the United States could afford to be without a radio receiving set.

Today the best available statistics indicate that 5,000,000 homes are equipped, and 21,000,000 homes remain to be supplied.

Radio receiving sets of the best reproductive quality should be made available for all, and we hope to make them cheap enough so that all may buy.

The day has gone by when the radio receiving set is a plaything. It must now be an instrument of service.

WEAF Purchased for $1,000,000

The Radio Corporation of America, therefore, is interested, just as the public is, in having the most adequate programs broadcast. It is interested, as the public is, in having them comprehensive and free from discrimination.

Any use of radio transmission which causes the public to feel that the quality of the programs is not the highest, that the use of radio is not the broadest and best use in the public interest, that it is used for political advantage or selfish power, will be detrimental to the public interest in radio, and therefore to the Radio Corporation of America.

To insure, therefore, the development of this great service, the Radio Corporation of America has purchased for one million dollars station WEAF from the American Telephone and Telegraph Company, that company having decided to retire from the broadcasting business.

The Radio Corporation of America will assume active control of that station on November 15.

National Broadcasting Company Organized

The Radio Corporation of America has decided to incorporate that station, which has achieved such a deservedly high reputation for the quality and character of its programs, under the name of the National Broadcasting Company, Inc.

The Purpose of the New Company

The purpose of that company will be to provide the best program available for broadcasting in the United States.

The National Broadcasting Company will not only broadcast these programs through station WEAF, but it will make them available to other broadcasting stations throughout the country so far as it may be practicable to do so, and they may desire to take them.

It is hoped that arrangements may be made so that every event of national importance may be broadcast widely throughout the United States.

No Monopoly of the Air

The Radio Corporation of America is not in any sense seeking a monopoly of the air. That would be a liability rather than an asset. It is seeking, however, to provide machinery which will insure a national distribution of national programs, and a wider distribution of programs of the highest quality.

If others will engage in this business the Radio Corporation of America will welcome their action, whether it be cooperative or competitive.

If other radio manufacturing companies, competitors of the Radio Corporation of America, wish to use the facilities of the National Broadcasting Company for the purpose of making known to the public their receiving sets, they may do so on the same terms as accorded to other clients.

The necessity of providing adequate broadcasting is apparent. The problem of finding the best means of doing it is yet experimental. The Radio Corporation of America is making this experiment in the interest of the art and the furtherance of the industry.

A Public Advisory Council

In order that the National Broadcasting Company may be advised as to the best type of program, that discrimination may be avoided, that the public may be assured that the broadcasting is being done in the fairest and best way, always allowing for human frailties and human performance, it has created an Advisory Council, composed of twelve members, to be chosen as representative of various shades of public opinion, which will from time to time give it the benefit of their judgment and suggestion. The members of this Council will be announced as soon as their acceptance shall have been obtained.

M. H. Aylesworth to be President

The President of the new National Broadcasting Company will be M. H. Aylesworth, for many years Managing Director of the National Electric Light Association. He will perform the executive and administrative duties of the corporation.

Mr. Aylesworth, while not hitherto identified with the radio industry or broadcasting, has had public experience as Chairman of the Colorado Public Utilities Commission, and, through his work with the association which represents the electrical industry, has a broad understanding of the technical problems which measure the pace of broadcasting.

One of his major responsibilities will be to see that the operations of the National Broadcasting Company reflect enlightened public opinion, which expresses itself so promptly the morning after any error of taste or judgment or departure from fair play.

We have no hesitation in recommending the National Broadcasting Company to the people of the United States.

It will need the help of all listeners. It will make mistakes. If the public will make known its views to the officials of the company from time to time, we are confident that the new broadcasting company will be an instrument of great public service.

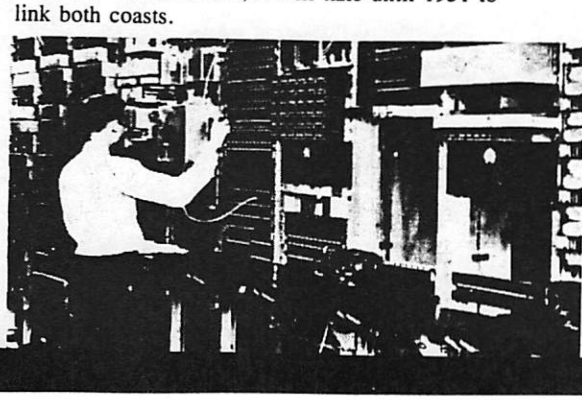
RADIO CORPORATION OF AMERICA

OWEN D. YOUNG, *Chairman of the Board* JAMES G. HARBORD, *President*

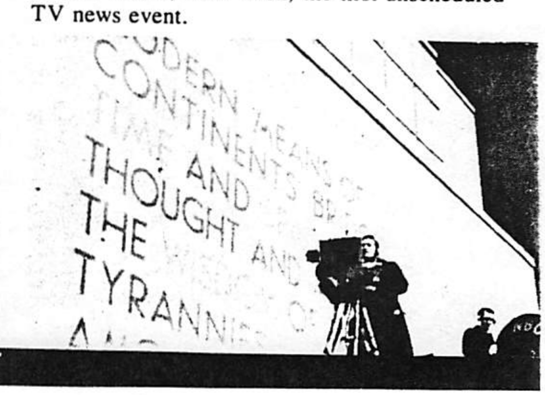
This public advertisement announced the formation of the National Broadcasting Company by RCA in 1926.

NBC places a TV antenna atop the Empire State Building in 1931. Picture definition improves dramatically.

1936. Bell Telephone engineers construct a coaxial cable between New York and Philadelphia with the capacity to carry simultaneous electrical impulses of different frequencies. In 1948, the East and Midwest will be linked; it will take until 1951 to link both coasts.



1937. NBC initiates the first mobile TV unit. It uses a microwave transmitter in a truck to relay images to the Empire State Building. In 1938, this unit will unexpectedly come upon and film a fire at Ward's Island, New York, the first unscheduled TV news event.



In 1930, David Sarnoff, a Russian immigrant, was elected president of the Radio Corporation of America.

The FCC chose the CBS system, and in 1950 gave the go-ahead for color on this basis. RCA sued, but lost, and in July 1951 CBS went on the air with color transmission. The industry did not follow. It did not like the idea of a color wheel in every receiver, and it wanted compatibility. Moreover, by this time RCA had developed the tri-color kinescope. In a few months CBS gave up on the field sequential system.

Meantime the industry formed another committee—the second NTSC (National Television Standards Committee). It deliberated for many months and eventually recommended a set of standards based on the RCA system (with some changes suggested by other manufacturers). On December 17, 1953, the FCC approved the start of color television broadcasting based on the NTSC-recommended standards.

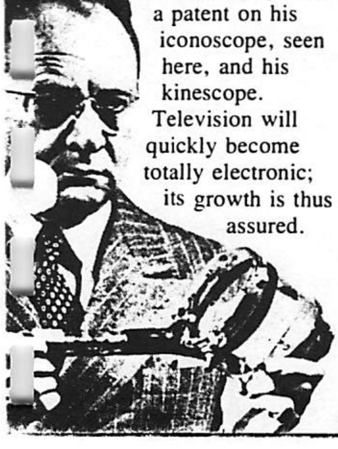
Color Telecasting Begins

It was the final approval needed. NBC began telecasting many of its programs in color. RCA started selling cameras and transmitting equipment to television stations, and color television receivers to the public. Other manufacturers soon followed. But color TV did not explode as monochrome had earlier. It turned out that producing programs in color was much more expensive than in monochrome. With only a few color receivers in use, advertisers were reluctant to pay the extra cost. And the public was slow to buy color receivers when only a few programs were in color. It was a classic chicken-and-egg situation. For awhile it looked as if color would fail. But RCA persevered, pouring millions of dollars into color promotion and subsidizing, in part, the extra cost of color production on NBC and on independent stations. Slowly the number of color receivers rose. By 1962 there were about one million color sets in use—sufficient to make the extra cost of color worthwhile to the advertiser. By 1965 there were five million color sets, and all the networks had gone to full-color. Color bloomed! By 1970 there were 37 million color sets.

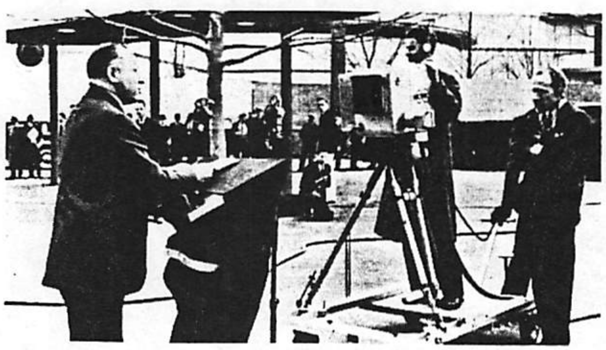
Further Improvements

Since 1953 there have been no further changes in the system used in the United States. But there have been tremendous changes in both home receivers and telecasting equipment. Increasing use of solid-state components (transistors, diodes) and of integrated circuits has made today's receivers smaller, lighter, more reliable—and they require far less power.

In telecasting equipment there have been corresponding advances. For example, RCA's 1954 color camera used 35 tubes, weighed 280 pounds and required 375 watts of power. RCA's newest comparable camera (studio type) uses just 3 tubes (the pickup

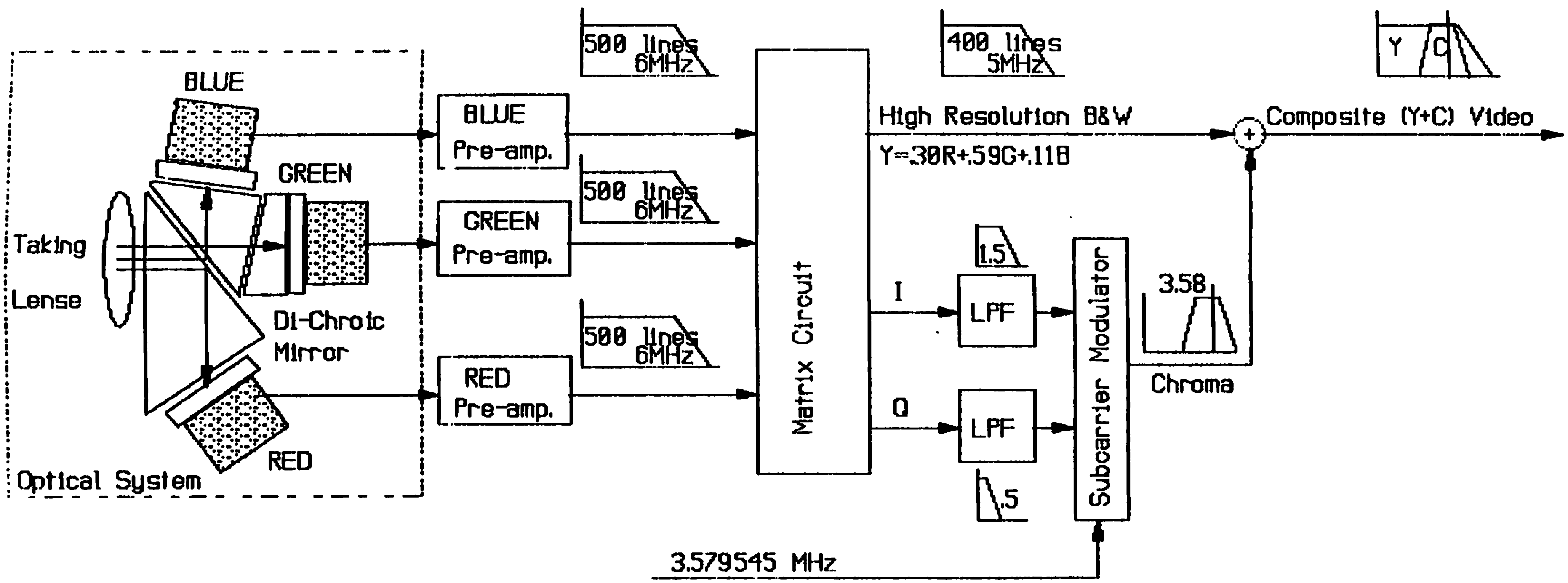


January 1, 1939. Vladimir Zworykin is granted a patent on his iconoscope, seen here, and his kinescope. Television will quickly become totally electronic; its growth is thus assured.



April 20, 1939. RCA President David Sarnoff dedicates the RCA pavilion at the New York World's Fair in the first scheduled news event covered by television. In his speech, Sarnoff predicts a great future for the new medium.

April 30, 1939. President Franklin D. Roosevelt speaks at the World's Fair, the first head of state to appear on television.


Optical System
Taking
Lense
BLUE
GREEN
RED
Di-Chroic
Mirror
BLUE
Pre-amp.
GREEN
Pre-amp.
RED
Pre-amp.
500 lines
6MHz
500 lines
6MHz
500 lines
6MHz
Matrix Circuit
400 lines
5MHz
High Resolution B&W
Y=.30R+.59G+.11B
I
Q
LPF
LPF
1.5
.5
Subcarrier Modulator
3.58
Chroma
Y
C
Composite (Y+C) Video
3.579545 MHz

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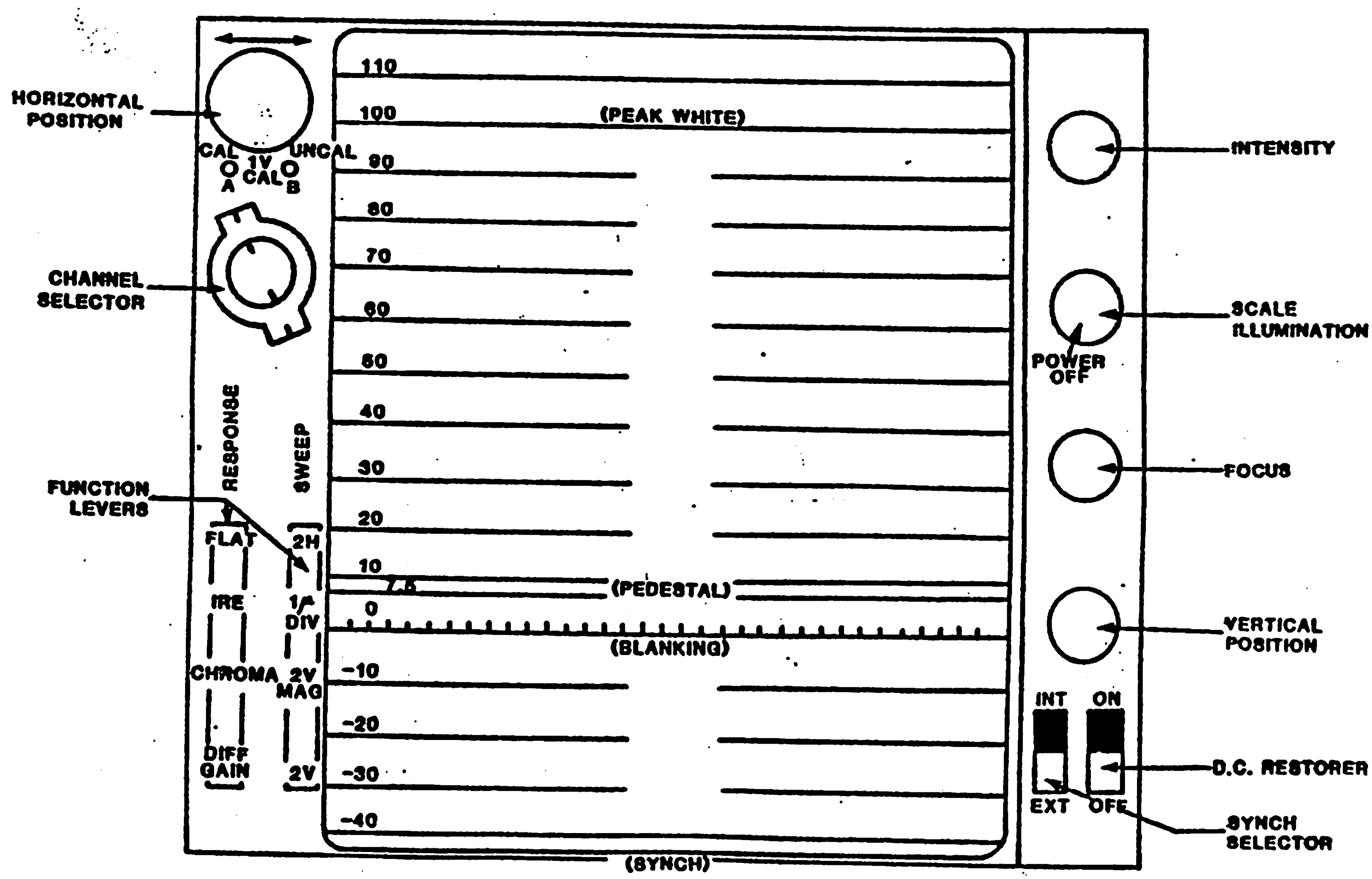
(9)

THE TEKTRONIX #528 TELEVISION WAVEFORM MONITOR



We can measure amplitude with waveform monitors. The first step is to assure that the input signal is properly terminated. Next, confirm that the vertical gain is in the "calibrated" and 1 volt display mode. Next check the calibration of the waveform monitor by selecting the CAL pulse. This pulse should go from 100 to -40, it is legal to adjust the vertical position for convenient viewing. Now that the reference is confirmed, voltage is measured by selecting a video input, and using the calibrated scales on the waveform monitor. Sync is measured from 0 to -40 as we stated before. Video is measured from 0 to 100, remember that active black video is at 7.5. Composite video is then measured from -40 to 100. There are some extra markings on the upper left portion of the graticule from 90 to 110. These are indicators of 2 IRE per division and are there for more accurate measurements.

Time measurements are made along the horizontal axis and use the markings on the "0" line. These markings represent 1 microsecond per major division in the 1 microsecond horizontal display mode. Oddly enough, they also represent 0.5 microseconds per major division in the 0.5 microsecond horizontal display mode. Time measurements then usually are made by selecting the 0.5 horizontal mode, and using the horizontal position control to position the measured part of the signal to a convenient marking for the measurement.

The last set of markings on the graticule are to measure K factor. A quick explanation is to consider K factor as one measurement which looks at signal distortion in against voltage changes. There are two arrows with adjacent side markers located about in the center of the graticule. These are markers used to position the special test signal rising and falling edges. The test signal is called a window and goes from 0 to 100. It is usually 18 microseconds wide so both the rising and falling edges overlay the vertical markers by the arrows. The box with the dotted lines at the 100 line above these markers is used to make the K factor measurement. The measurements are made in the box only, not before it. The window might have overshoots or ringing before entering the box, but how it looks in the box is all that's measured. The standard window is 18 microseconds and the box is such that the first and last microsecond are not measured. If the window is inside the dotted lines, then it is less than 2% K factor, the outside box indicates 4%. Any window may be used, no matter how long the window is, just set the rising edge on the left marker and make assessments in the box.

Now use the waveform monitor manual and try to understand the instrument a bit better. I hope that these last three issues are some help. You may always call or write Mark Everett at Videotek with your questions, requests and comments.

July, 1989
No. 14

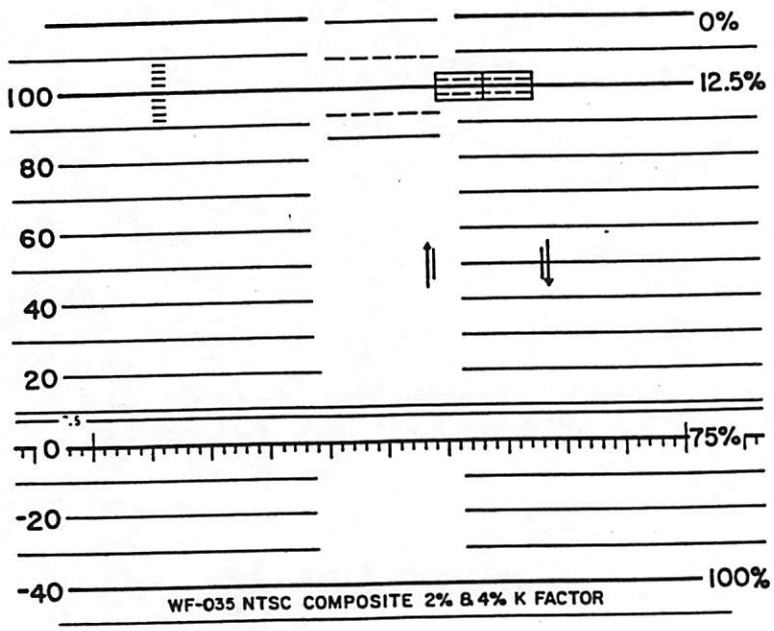
APPLICATION

VIDEOTEK INC.

243 Shoemaker Road Pottstown Pennsylvania 19464 / (215) 327-2292 TWX 710-653-0125 FAX (215) 327-9295

Using A Waveform Monitor

The previous two issues have dealt with the connections, knobs and switches on waveform monitors. This issue will address just what you see and how to interpret the information. Let's begin with a simple view of the CRT (Cathode Ray Tube) graticule. This is from our TSM line of waveform monitors, and this particular graticule is for NTSC use.



The horizontal lines are used to measure signal (voltage) amplitude. They are marked from 100 on the top thru 0 to -40 at the bottom. There are hefty lines at 120, 100, 0 and -40. All of these heavier lines are important markers for certain measurements. We have major lines every 10 units, and some special lines placed other places. The markings on the left side (100, 80, etc.) are equal divisions and are called IRE or Institute of Radio Engineers divisions. In regular baseband video, a normal 1 volt signal of video is 140 IRE units. The markings on the right side are from 0% on the top to 100% on the bottom. These markers are used to measure depth of modulation, and have no application in anything but R.F. broadcast applications. The extra full length horizontal line at 7.5 IRE is the NTSC set-up reference. The rest of the markings have special applications and will all be treated one at a time.

A typical composite video signal will extend from the -40 markers to the 100 markers. I will always refer to the left side (IRE) markers. I just know that someone is looking at their waveform monitor and seeing other strange things. So let's all get together and apply color bars to the input, naturally we all know to terminate the signal. Then let's look at the signal in the 1 H horizontal display mode. Finally, remove some of the confusing information by turning on the low pass (IRE) filter. Now, the burst is not visible, but the sync and video are clearly displayed. Sync pulses are the (usually) lowest part of the video signal, going from 0 to -40. Color burst on a composite signal will go from 20 to -20. Video, at least the luminance portion, should be no greater than 100 nor less than 7.5. Just for your information, 7.5 is black and 100 is white. Certain situations will allow video to go as high as 120, but anything higher will most certainly cause problems somewhere. Black, likewise, is sometimes below 7.5 but again too much will cause certain problems. Let the color information back on the display by pressing the Flat filter button. Now we have video information higher than 100 and as low as -20. This is color information and it is at those "illegal" levels for a very short period of time.

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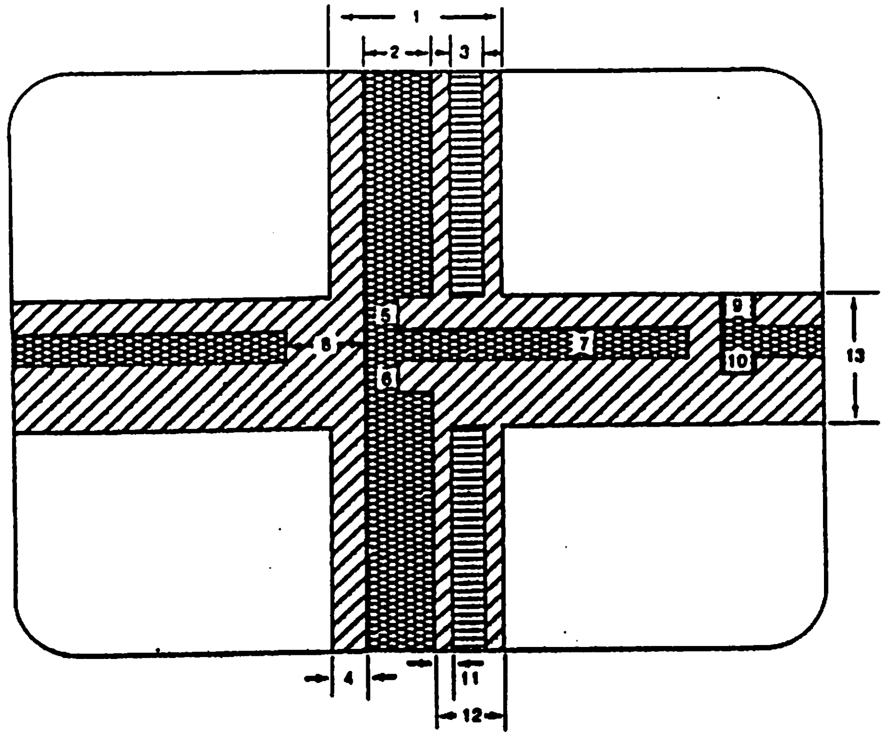
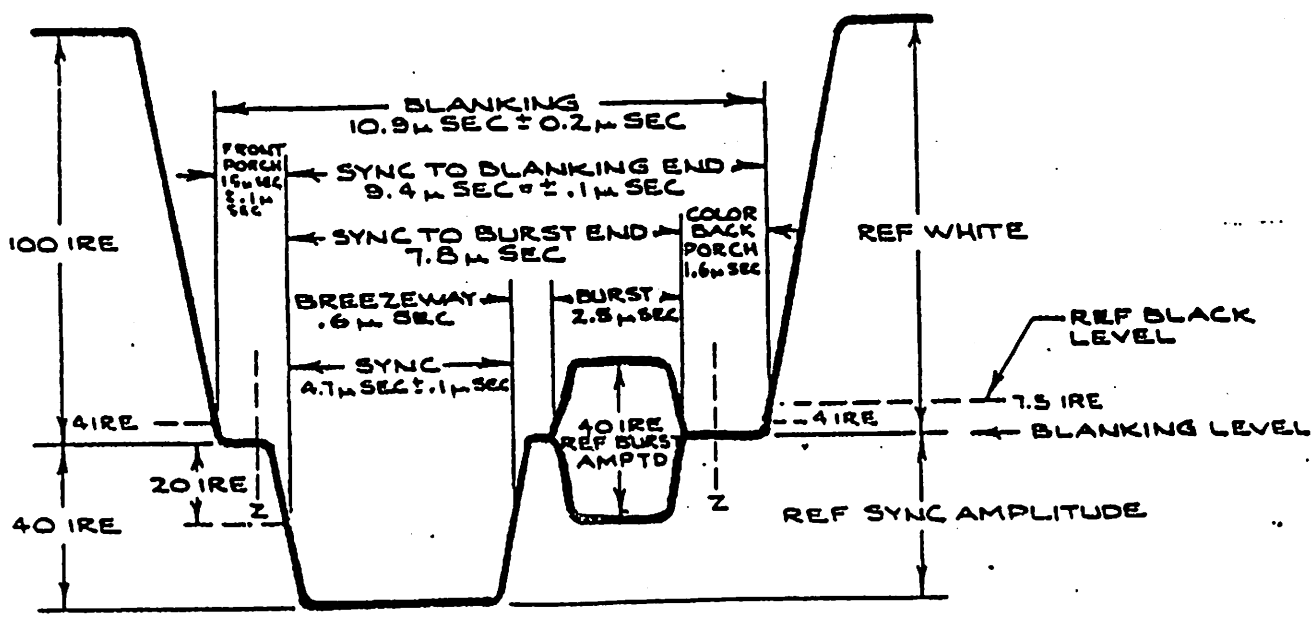
continued . . . .



1. Horizontal Blanking
2. Horizontal Sync
3. Color Burst (if present)
4. Front Porch
5. Odd Leading Equalizing Pulses
6. Odd Trailing Equalizing Pulses
7. Odd Vertical Sync
8. Vertical Serrations
9. Even Leading Equalizing Pulses
10. Even Trailing Equalizing Pulses
11. Breezeway
12. Back Porch
13. Vertical Blanking



HORIZONTAL BLANKING

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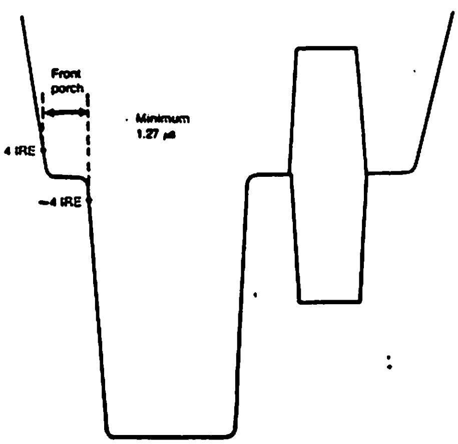
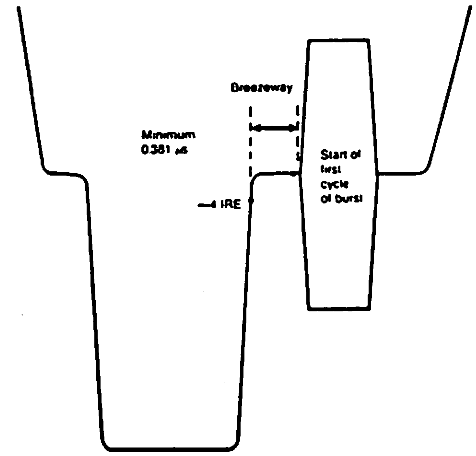
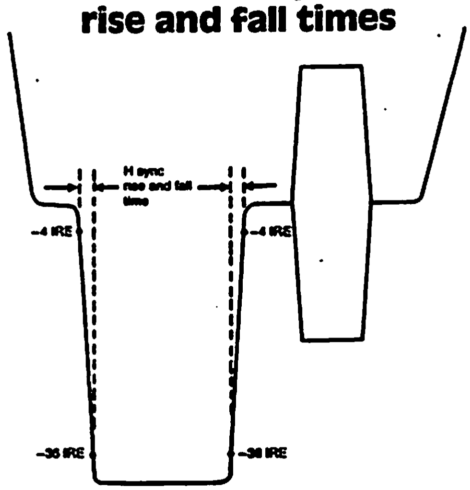
Front porch



Breezeway

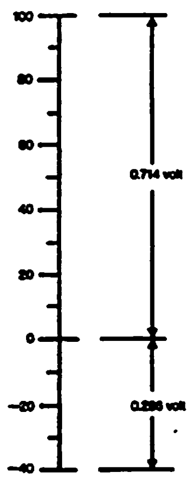


Horizontal sync rise and fall times



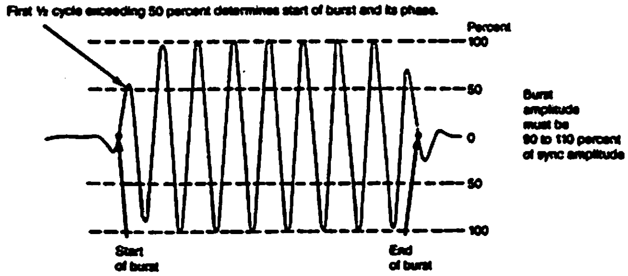
The rise and fall times of horizontal sync—measured between the 10 and 90 percent points on the pulse leading and trailing edges—are not to exceed 0.250 microsecond.

IRE scale units relative to 1 volt peak-to-peak video amplitude



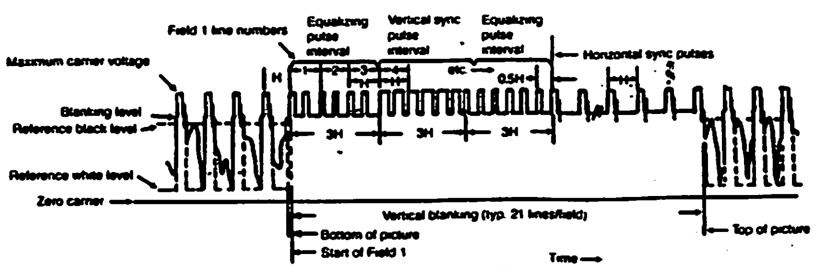
1 IRE unit = 0.0071429 volt

Burst characteristics



Specification for burst requires 8 to 11 cycles

Vertical interval



Serrations in vertical sync pulse must each be 3.8 to 5.1 microseconds in width.
Equalizing pulse width should be 2.0 to 2.54 microseconds.

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There are two basic types of NTSC color bar signals in common use. The terms "75% bars" and "100% bars" are generally used to distinguish between the two types. While this terminology is widely used, there is often confusion about exactly which parameters the 75% versus 100% notation refers to.

RGB Amplitudes

The 75%/100% nomenclature specifically refers to the maximum amplitudes reached by the Red, Green and Blue signals when they form the six primary and secondary colors required for color bars. For 75% bars, the maximum amplitude of the RGB signals is 75% of the peak white level. For 100% bars, the RGB signals can extend up to 100% of peak white. See Figures 110 and 111.

Saturation

Both 75% and 100% amplitude color bars are 100% saturated. In the RGB format, colors are saturated if at least one of the primaries is at zero. Note in Figures 110 and 111 that the zero signal level is at setup (7.5 IRE) for NTSC.

The Composite Signal

In the composite signal, both chrominance and luminance amplitudes vary according to the 75%/100% distinction. However, the ratio between chrominance and luminance amplitudes remains constant in order to maintain 100% saturation. See Figures 112 and 113 on page 74.

White Bar Levels

Color bar signals can also have different white bar levels, typically either 75% or 100%. This parameter is completely independent of the 75%/100% amplitude distinction, and either white level may be associated with either type of bars.

Effects of Setup

Because of setup, the 75% signal level for NTSC is at 77 IRE. The maximum available signal amplitude is 100 –7.5, or 92.5 IRE. 75% of 92.5 IRE is 69.4 IRE, which when added to the 7.5 IRE pedestal yields a level of approximately 77 IRE.

Note in the Figure 110 that the 75% white bar and the 75% RGB signals extend to 77 IRE.

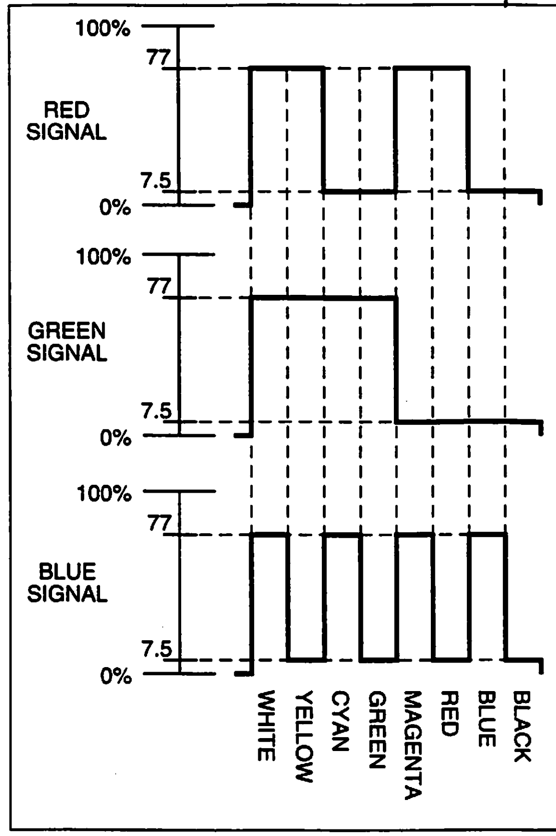


Figure 110. RGB levels decoded from 75% bars with 75% white.

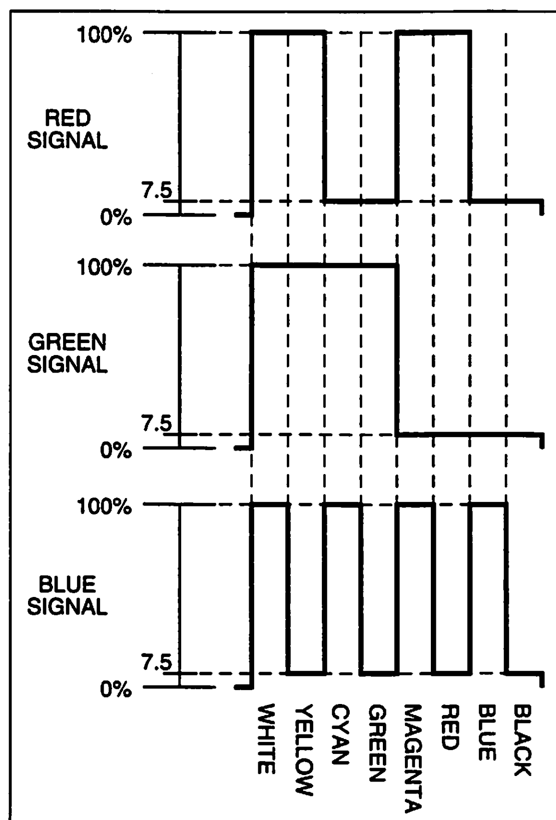
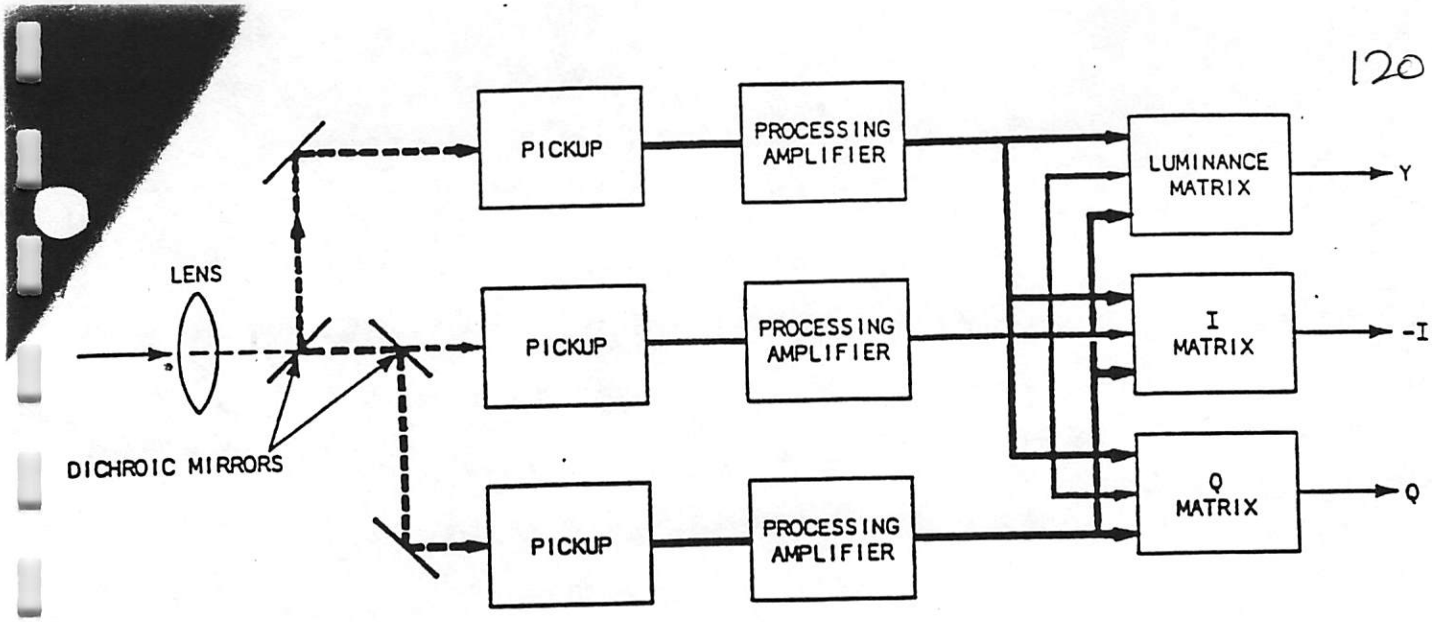
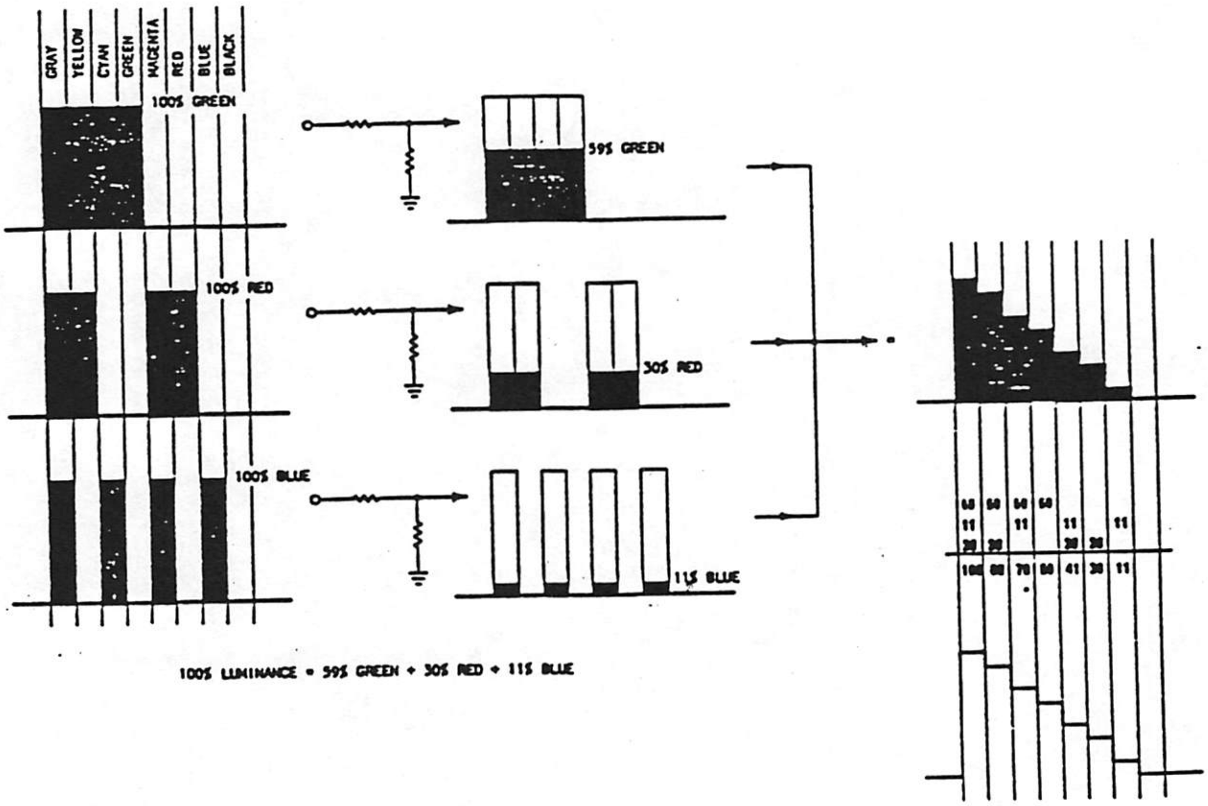


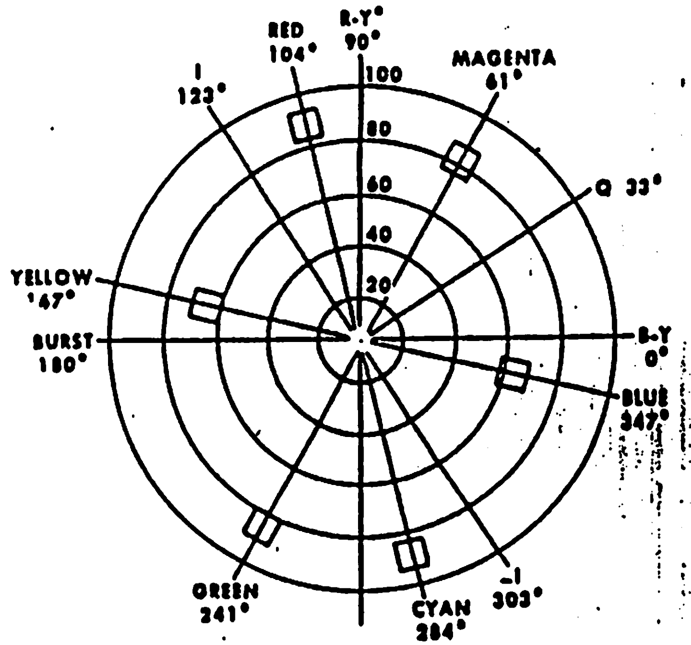
Figure 111. RGB levels decoded from 100% bars with 100% white.



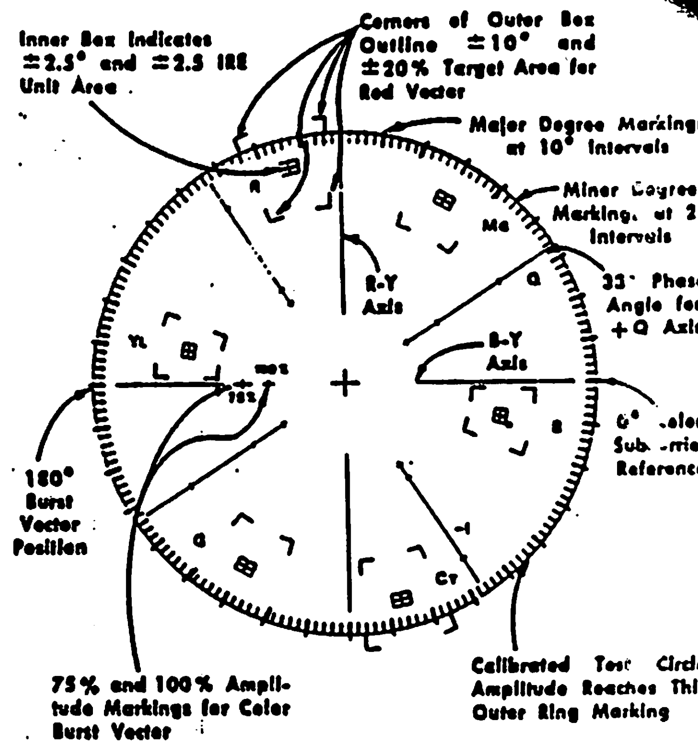
Block diagram of color transformation process.

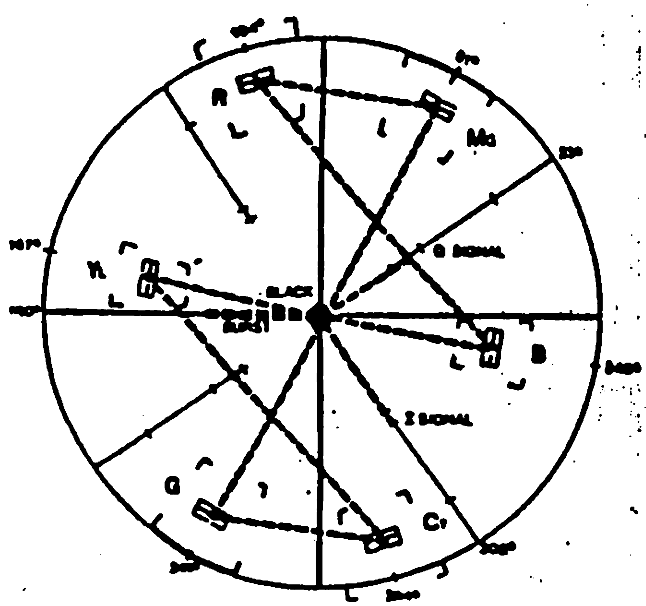


Graphic representation of the luminance matrix transformation from red, green and blue images.



NORMAL VECTOR DISPLAY-SPLIT FIELD COLOR BARS





VECTORSCOPE GRATICULE

AND DISPLAY

FIGURE #II-4

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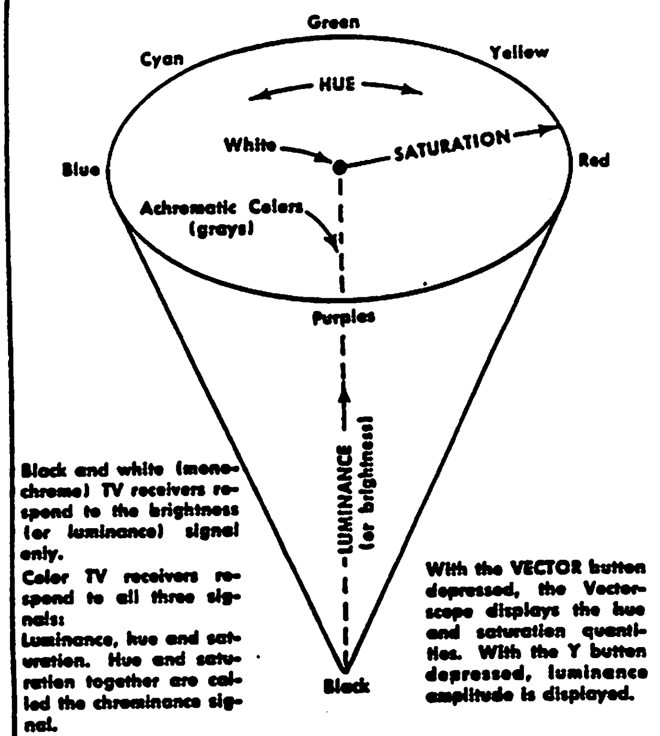
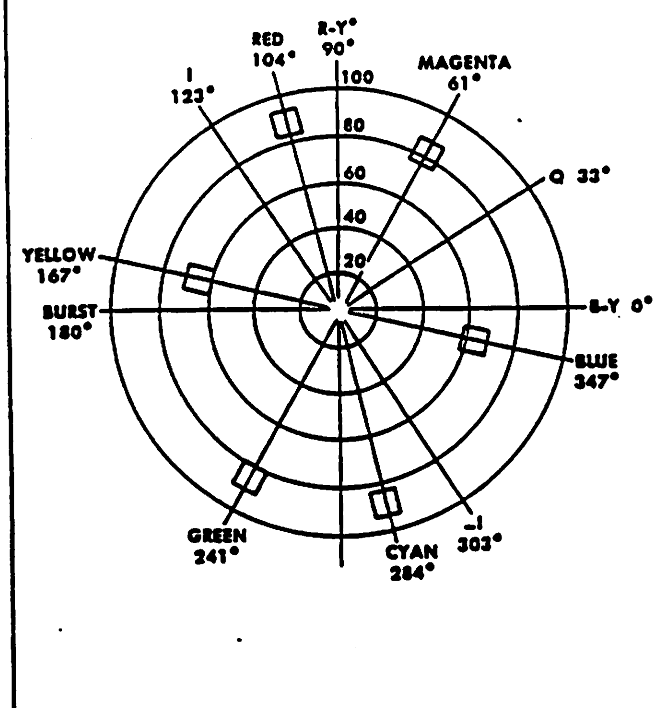
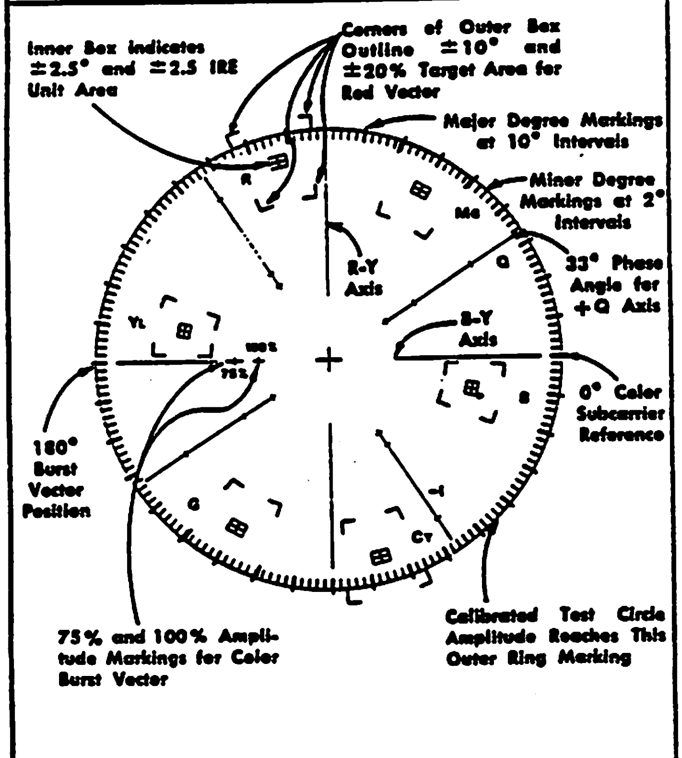


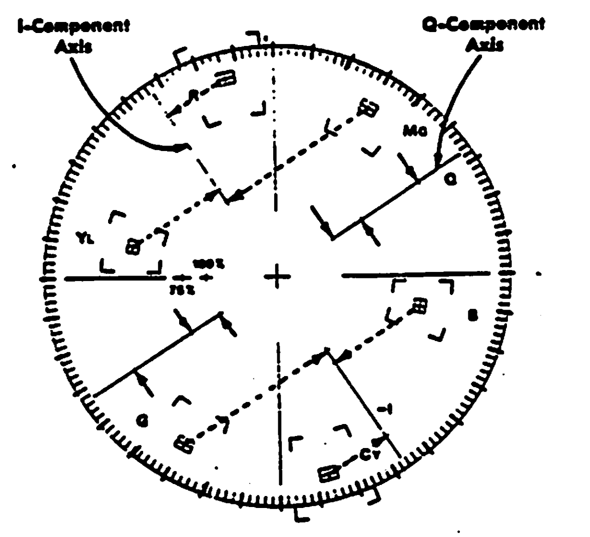
(A) Conical three-dimensional representation of color concepts.



(B) Standard color-phase vector diagram showing vector relationship among chrominance components.



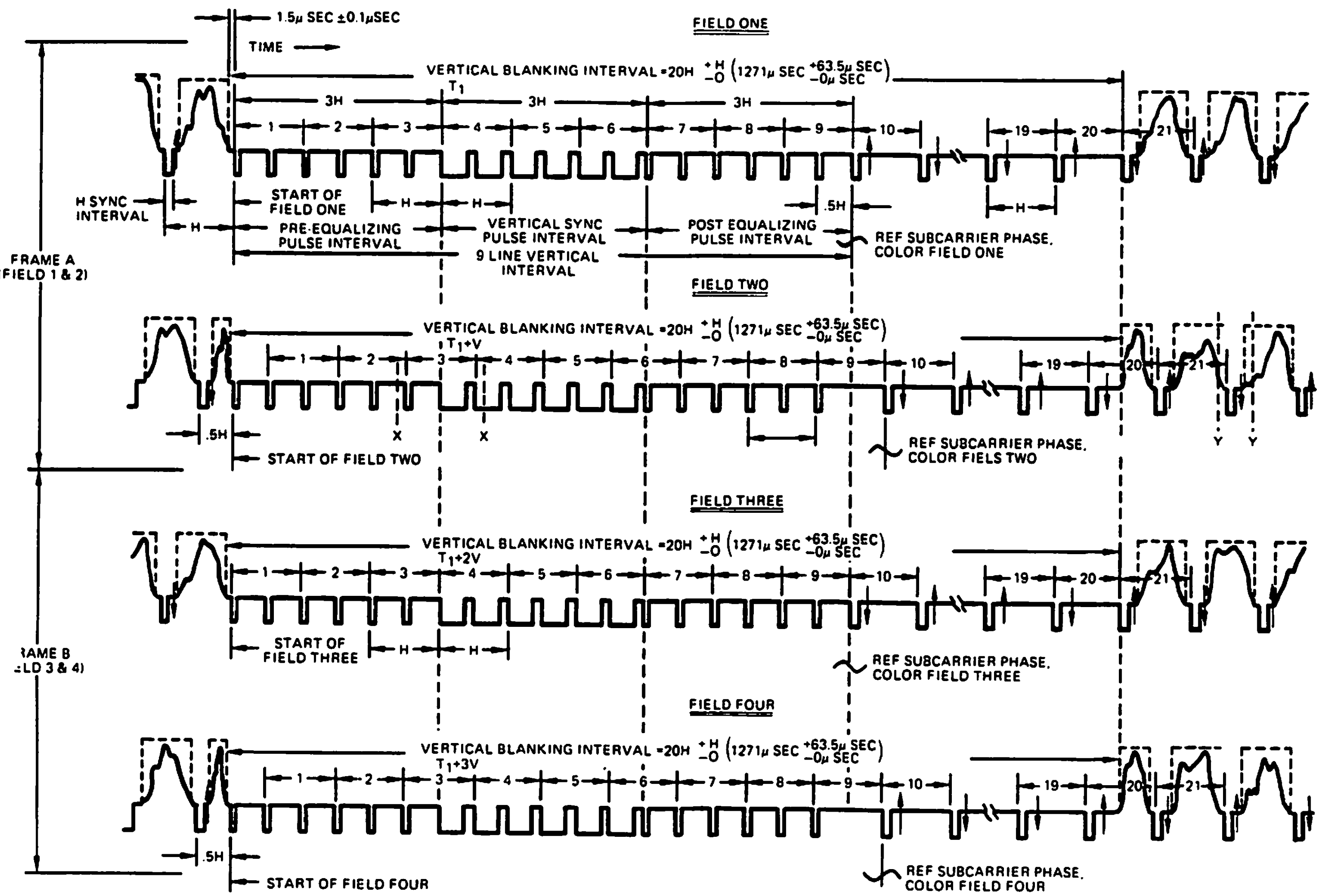
(C) Vector (internal) graticule on the Type R520 with color vector and phase markings defined to aid in interpreting vector displays.

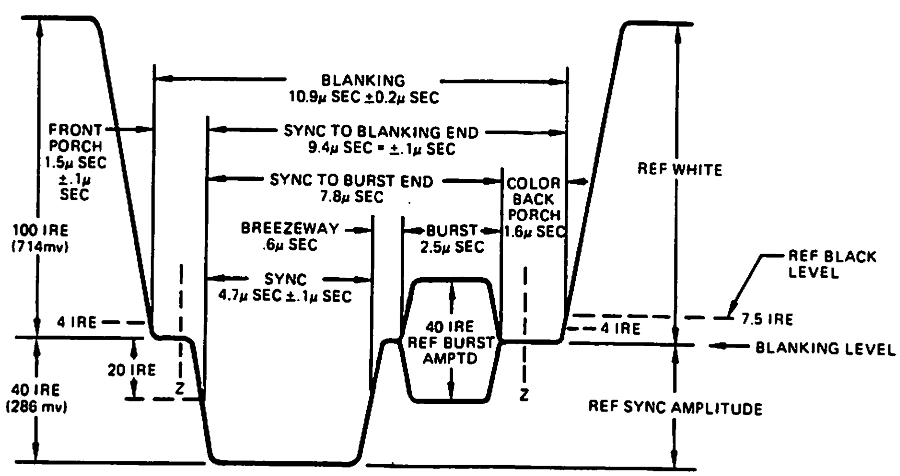


Dashed line with arrow associates the color vectors with the I-axis markings. If Q component of the signal is absent, linear color vectors will move to their associated markings on the I-axis.

Short arrows along the Q-axis are intended to show relationship between Q-axis markings and the associated color vectors. If I component in the signal is absent, linear color vector points will appear at the appropriate marks along the Q axis.

(D) Vector graticule on the Type R520 showing purpose of I and Q axis markings.


1.5μ SEC ±0.1μSEC
TIME
FIELD ONE
VERTICAL BLANKING INTERVAL =20H +H -O (1271μ SEC +63.5μ SEC -0μ SEC)
T1
3H
H SYNC INTERVAL
START OF FIELD ONE
PRE-EQUALIZING PULSE INTERVAL
VERTICAL SYNC PULSE INTERVAL
POST EQUALIZING PULSE INTERVAL
9 LINE VERTICAL INTERVAL
.5H
REF SUBCARRIER PHASE, COLOR FIELD ONE
FRAME A (FIELD 1 & 2)
FIELD TWO
T1+V
START OF FIELD TWO
REF SUBCARRIER PHASE, COLOR FIELS TWO
FIELD THREE
T1+2V
START OF FIELD THREE
REF SUBCARRIER PHASE, COLOR FIELD THREE
FRAME B (FIELD 3 & 4)
FIELD FOUR
T1+3V
START OF FIELD FOUR
REF SUBCARRIER PHASE, COLOR FIELD FOUR


BLANKING 10.9μ SEC ±0.2μ SEC
FRONT PORCH 1.5μ SEC ±.1μ SEC
SYNC TO BLANKING END 9.4μ SEC = ±.1μ SEC
SYNC TO BURST END 7.8μ SEC
COLOR BACK PORCH 1.6μ SEC
REF WHITE
100 IRE (714mv)
BREEZEWAY .6μ SEC
BURST 2.5μ SEC
SYNC 4.7μ SEC ±.1μ SEC
REF BLACK LEVEL
4 IRE
7.5 IRE
40 IRE REF BURST AMPTD
BLANKING LEVEL
20 IRE
40 IRE (286 mv)
REF SYNC AMPLITUDE
Z

DETAIL YY

APPENDIX C — RS-170A

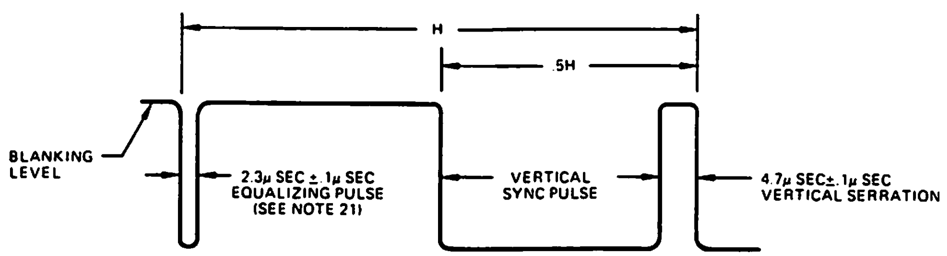
NTSC Standard

NOTES

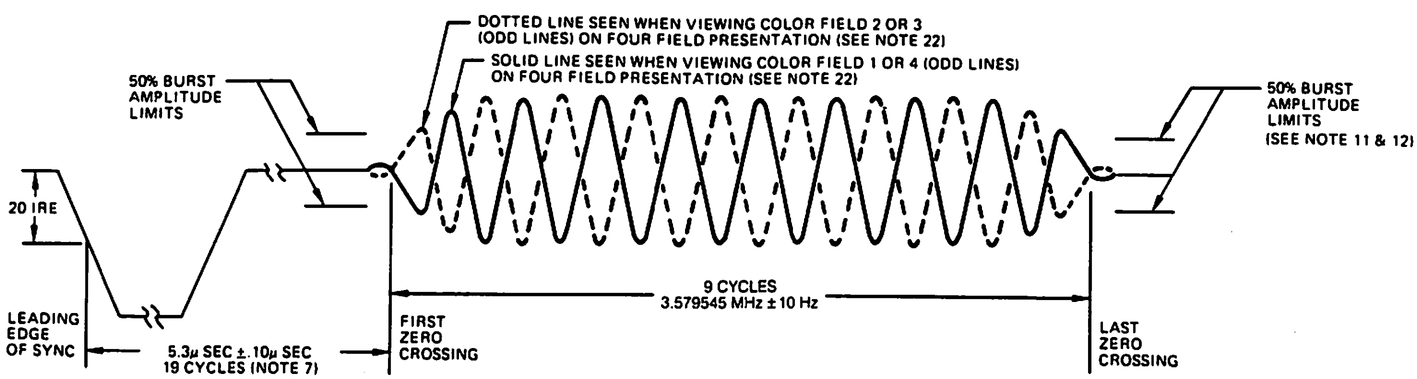
1. SPECIFICATIONS APPLY TO STUDIO FACILITIES. NETWORK AND TRANSMITTER CHARACTERISTICS ARE NOT INCLUDED.
2. ALL TOLERANCES AND LIMITS SHOWN IN THIS DRAWING PERMISSABLE ONLY FOR LONG TIME VARIATIONS.
3. BURST FREQUENCY SHALL BE 3.579545 MHz ±10 Hz
4. HORIZONTAL SCANNING FREQUENCY SHALL BE 2/455 TIMES THE BURST FREQUENCY
5. VERTICAL SCANNING FREQUENCY SHALL BE 2/525 TIMES THE HORIZONTAL SCANNING FREQUENCY
6. START OF COLOR FIELDS ONE AND THREE IS DEFINED BY A WHOLE LINE BETWEEN THE FIRST EQUALIZING PULSE AND THE PRECEDING H SYNC PULSE. START OF COLOR FIELDS TWO AND FOUR DEFINED BY A HALF LINE BETWEEN THE FIRST EQUALIZING PULSE AND THE PRECEDING H PULSE. COLOR FIELD ONE IS THAT FIELD WITH POSITIVE GOING ZERO CROSSINGS OF REFERENCE SUBCARRIER NOMINALLY COINCIDENT WITH 50% AMPLITUDE POINT OF THE LEADING EDGES OF EVEN NUMBERED HORIZONTAL SYNC PULSES.
7. THE ZERO CROSSINGS OF REFERENCE SUBCARRIER SHALL BE NOMINALLY COINCIDENT WITH THE 50% POINT OF THE LEADING EDGES OF ALL HORIZONTAL SYNC PULSES. FOR THOSE CASES WHERE THE RELATIONSHIP BETWEEN SYNC AND SUBCARRIER IS CRITICAL FOR PROGRAM INTEGRATION, THE TOLERANCE ON THIS COINCIDENCE IS ±45 OF REFERENCE SUBCARRIER
8. ALL RISE TIMES AND FALL TIMES UNLESS OTHERWISE SPECIFIED ARE TO BE 0.140 μSEC ±0.02 μSEC MEASURED FROM TEN TO NINETY PER CENT AMPLITUDE POINTS. ALL PULSE WIDTHS EXCEPT BLANKING ARE MEASURED AT FIFTY PER CENT AMPLITUDE POINT
9. OVERSHOOT ON ALL PULSES DURING SYNC AND BLANKING (VERTICAL AND HORIZONTAL) SHALL NOT EXCEED TWO IRE UNITS. ANY OTHER EXTRANEOUS SIGNALS DURING BLANKING INTERVALS SHALL NOT EXCEED TWO IRE UNITS, MEASURED OVER A BANDWIDTH OF 6 MHz.
10. BURST ENVELOPE RISE TIME IS 0.30 μSEC MEASURED BETWEEN THE TEN AND NINETY PERCENT AMPLITUDE POINTS. IT SHALL HAVE THE GENERAL SHAPE SHOWN
11. THE START OF BURST IS DEFINED BY THE ZERO CROSSING (POSITIVE OR NEGATIVE SLOPE) THAT PRECEEDS THE FIRST HALF CYCLE OF SUBCARRIER THAT IS 50% OR GREATER OF THE BURST AMPLITUDE
12. THE END OF BURST IN DEFINED BY THE ZERO CROSSING (POSITIVE OR NEGATIVE SLOPE) THAT FOLLOWS THE LAST HALF CYCLE OF SUBCARRIER THAT IS 50% OR GREATER OF THE BURST AMPLITUDE
13. MONOCHROME SIGNALS SHALL BE IN ACCORDANCE WITH THIS DRAWING EXCEPT THAT BURST IS OMITTED, AND FIELDS THREE AND FOUR ARE IDENTICAL TO FIELDS ONE AND TWO RESPECTIVELY
14. REFERENCE SUBCARRIER IS A CONTINUOUS SIGNAL WHICH HAS THE SAME INSTANTANEOUS PHASE AS BURST
15. PROGRAM OPERATING LEVEL WHITE IS 100 IRE, +0, −2 IRE
16. PROGRAM OPERATING LEVEL BLACK IS 7.5 IRE, ±2.5 IRE.
17. PROGRAM OPERATING LEVEL SYNC IS 40 IRE, ±2 IRE
18. PROGRAM OPERATING LEVEL BURST IS 40 IRE, ±2 IRE.
19. BURST PEDESTAL NOT TO EXCEED ±2 IRE.
20. BREEZEWAY, BURST, COLOR BACK PORCH, AND SYNC TO BURST END ARE NOMINAL IN DETAIL BETWEEN YY. SEE DETAIL BETWEEN ZZ FOR TOLERANCES.
21. RATIO OF AREA OF VERTICAL EQUALIZING PULSE TO SYNC PULSE SHALL BE WITHIN 45 TO 50 PER CENT
22. THERE WILL BE A 100 DEGREE REVERSAL OF PHASE WHEN VIEWING EVEN LINES ON A FOUR FIELD PRESENTATION. A FOUR FIELD PRESENTATION MEANS A DISPLAY DEVICE WHICH IS TRIGGERED BY FOUR FIELD (15 Hz) INFORMATION.

THIS DRAWING CORRESPONDS TO PROPOSED RS 170A VIDEO STANDARD.

COLOR TIMING DATA:
1° = .776 ns
INS = 1.289°
FOR CABLE WITH 66% PROPAGATION FACTOR,
1° = 6.035'' = .503'
INS = 7.778'' = .648'

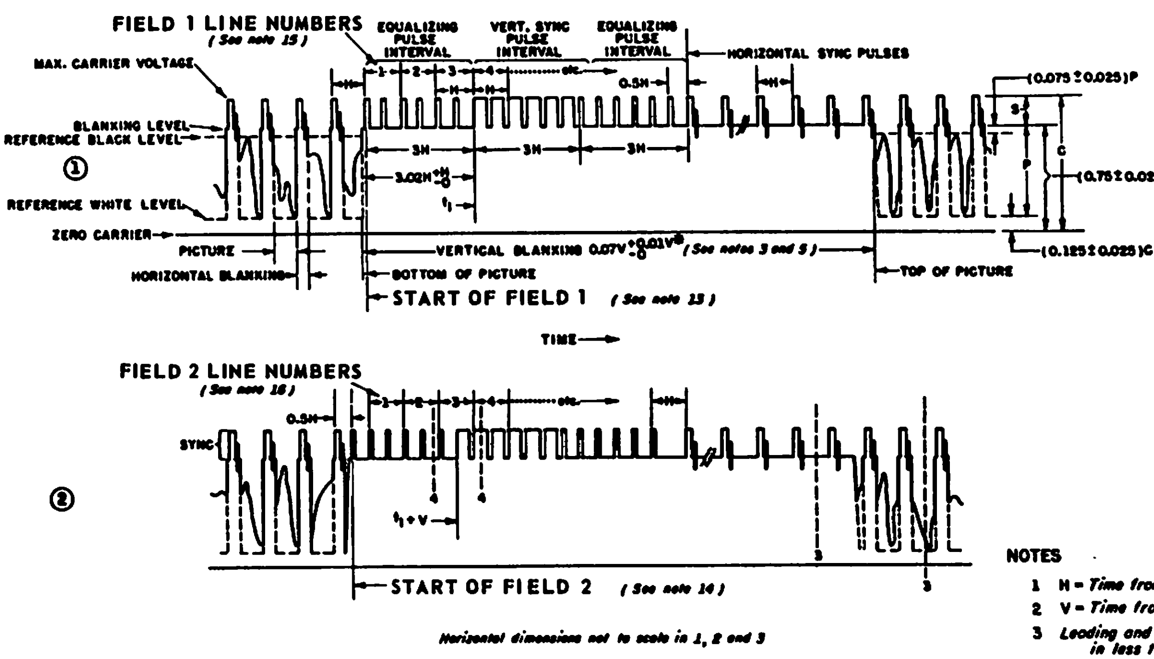


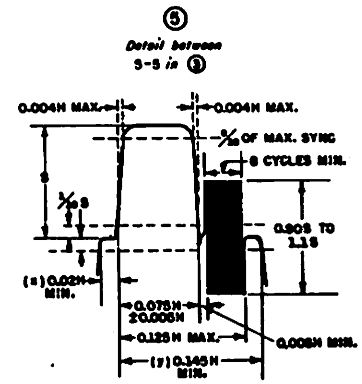
DETAIL XX

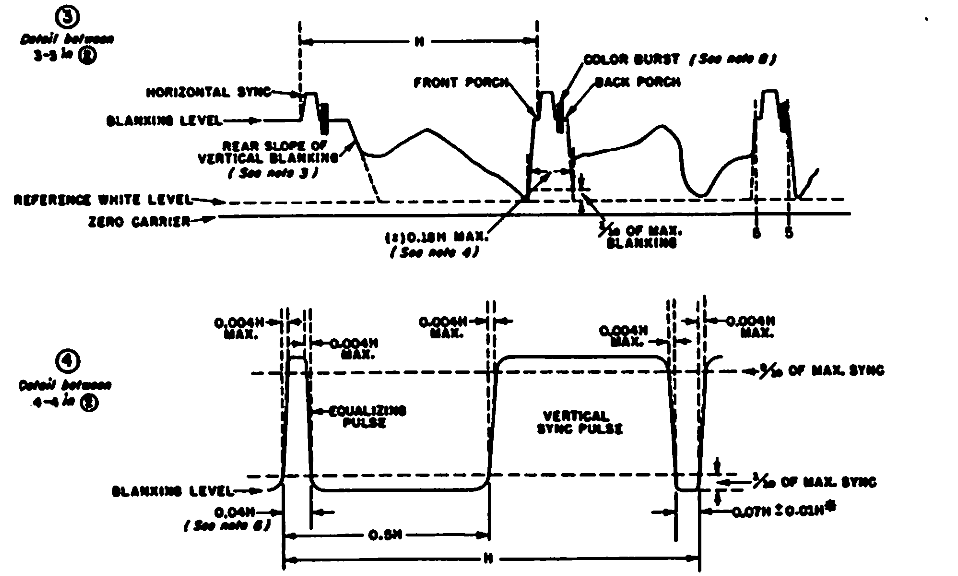


DETAIL ZZ

APPENDIX D — FCC 73.699, Fig. 6







NOTES

1 H = Time from start of one line to start of next line.

2 V = Time from start of one field to start of next field.

3 Leading and trailing edges of vertical blanking should be complete in less than 0.1H.

4 Leading and trailing slopes of horizontal blanking must be steep enough to preserve minimum and maximum values of (x+y) and (z) under all conditions of picture content.

*5 Dimensions marked with asterisk indicate that tolerances given are permitted only for long time variations and not for successive cycles.

6 Equalizing pulse area shall be between 0.45 and 0.5 of area of a horizontal sync pulse.

7 Color burst follows each horizontal pulse, but is omitted following the equalizing pulses and during the broad vertical pulses.

8 Color bursts to be omitted during monochrome transmission.

9 The burst frequency shall be 3.579545 mc. The tolerance on the frequency shall be ±10 cycles with a maximum rate of change of frequency not to exceed 1/10 cycle per second per second.

10 The horizontal scanning frequency shall be $\frac{2}{455}$ times the burst frequency.

11 The dimensions specified for the burst determine the times of starting and stopping the burst, but not its phase. The color burst consists of amplitude modulation of a continuous sine wave.

12 Dimension "P" represents the peak excursion of the luminance signal from blanking level, but does not include the chrominance signal. Dimension "S" is the sync amplitude above blanking level. Dimension "C" is the peak carrier amplitude.

13 Start of Field 1 is defined by a whole line between first equalizing pulse and preceding H sync pulses.

14 Start of Field 2 is defined by a half line between first equalizing pulse and preceding H sync pulses.

15 Field 1 line numbers start with first equalizing pulse in Field 1.

16 Field 2 line numbers start with second equalizing pulse in Field 2.

17 Refer to text for further explanations and tolerances.

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Section 3

SC/H Phase: Problems & Solutions

Definition of SC/H Phase

In the late 1940s the Electronic Industries Association (EIA) established monochrome television standard RS-170. In recent years proposed color standard RS-170A has received increasing acceptance. RS-170A fully outlines the phase relationship of the color subcarrier to horizontal sync. A graphic representation of this standard is included on pages 12 - 13. If we look at the equation that relates horizontal sync to subcarrier and consider the number of lines in each frame, several conclusions can be made.

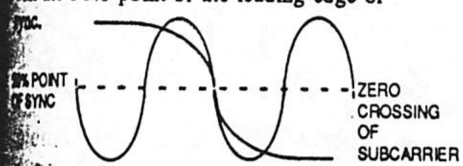
?

$$H = \frac{2 \times 3579545}{455}$$

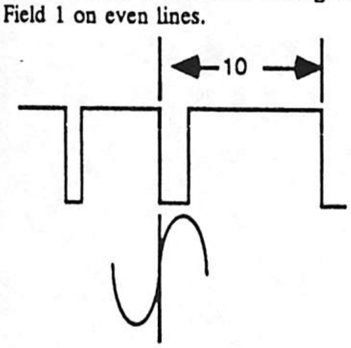
First, there are 227.5 subcarrier cycles per horizontal line, so subcarrier phase reverses every line. This is desirable to reduce the visibility of color subcarrier on monochrome receivers. Second, with 525 lines per frame, there are 119437.5 subcarrier cycles each frame. This causes subcarrier phase to reverse every frame. Because of the extra half cycle of subcarrier, it takes two frames to complete one full four field color sequence, called a color frame. It is clear from the horizontal frequency equation above that horizontal is frequency locked to subcarrier, but it does not define the phase relationship between them. Proposed color standard RS-170A clearly defines **SC/H phase** as: the zero crossing of the extrapolated subcarrier of color burst shall align with the fifty percent point of the leading edge of horizontal sync. For color field one, the extrapolated subcarrier zero crossing will be **positive** going on even lines. This definition of sync to subcarrier phase (SC/H) is required for the unambiguous identification of the four field color sequence. The operational ramifications of these definitions are not obvious and require further explanation.

C/H Phase

Alignment of the zero crossing of subcarrier with the 50% point of the leading edge of sync.



Reference subcarrier is **Positive Going** in Color Field 1 on even lines.



Operational Importance Of SC/H Phase

The importance of SC/H phase lies primarily in the video tape editing environment. If during playback the video signal coming off the tape is not of the same color frame as the house reference, the video at the machine's time base corrector output must be shifted horizontally. The shift can be in either direction and be up to 140 nonoseconds (one half subcarrier cycle). This may result in loss of active picture and a widening of blanking since the output processor blanking is referenced to the house. Even if the off-tape video is of the correct color frame, the machine-output video will be shifted horizontally to a smaller degree in an amount equal to any SC/H phase difference between the off-tape and house video.

These horizontal shifts are troublesome in a tape editing environment, especially when editing scenes together of similar content. At the edit point the background will appear to jump horizontally. This is unacceptable and thus dictates the need for an entirely SC/H phased facility.

To insure the proper operation of the tape machine color framing circuits (to avoid incorrect color frame operation), the SC/H phase relationship of the video recorded on tape and house video must match. As a matter of uniformity correct SC/H phase is defined by RS-170A. It is important that all recorded video have a constantly correct SC/H phase relationship. The reference input to the tape machine should also be a stable SC/H phase source.

Problems Achieving and Maintaining SC/H Phase

Subcarrier timing in a studio is a well understood concept in the industry; if it is not correct, there will be color hue shifts between sources. If sync timing is not correct, horizontal shifts will occur at the video switcher. The concept of SC/H phasing in a studio requires a higher level of thought regarding each element within the studio.

First, and most obvious is the house sync generator. If the sync generator cannot generate consistent SC/H phased outputs, maintaining SC/H phase in the plant will never be possible. It is equally important that all the sync generators in a multiple sync generator facility maintain correct SC/H phase and color frame relationships.

Once SC/H phase has been defined by the sync generator none of the elements in the system should alter the SC/H phase. Some elements are obvious like the video processor which regenerates sync and burst. If the phase of the regenerated sync or burst is different from the incoming video, the SC/H phase is altered. Less obvious are sources which derive timing from externally applied sync and subcarrier. If sync and subcarrier are fanned out through DAs, then their phase can be altered independently. This dictates the output of each source device be SC/H phased prior to or at the input of the switcher. There are many distortions which make the determination of color frame and SC/H phase difficult. The most prominent is sync to subcarrier time base error. This can be generated by many devices, such as sync generators with noise in the horizontal sync circuits, **linear and regenerative pulse DAs** which suffer from **pick-off jitter** or low frequency response problems, or any device that has separate sync and subcarrier regeneration circuitry.

Noise, low-frequency smear, hum, and power glitches are distortions that may occur in signal transmissions. If these are not removed prior to sync separation, determination of the exact fifty percent point of sync will be difficult.

Linear and Regenerative Pulse DAs
Linear pulse DA will handle 4V p-p signals (pulses) but is limited to amplifying and fanning out the signal. Regenerative pulse DA reconstructs the signal and allows for adjustment of delay.

Pick-Off Jitter
Jitter is a random aberration in the time period due to noise or time base instability. Pick-off means sample point.

Measuring SC/H Phase

The SC/H (subcarrier-to-horizontal) phase is the time relationship between the subcarrier and the leading edge of horizontal sync. A properly adjusted SC/H phase occurs when the 50% points of the leading edge of sync and the subcarrier zero crossings are coincident.

The color frame pulse (V1) appears on line 11 of field 1. V1 identifies field 1 of the 4 field color sequence.

Test Equipment Required

The following test equipment is required to perform the SC/H phase measurement procedure. Equivalent test equipment may be substituted but must be equal to or superior in performance.

Dual Trace Oscilloscope (with delayed sweep and one channel input inversion)	Tektronix 465
Switchable Delay Line or Subcarrier Delay DA (360° range)	Mathey 511

Test Procedure

SC/H Phase Measurement

1. Connect a video source requiring SC/H phase measurement to the inverting channel of the oscilloscope.

2. Connect subcarrier (3.58 MHz continuous) to the second channel of oscilloscope.

3. While observing the oscilloscope (triggered at a horizontal rate), adjust subcarrier to match amplitude of burst.

4. At the oscilloscope, invert the video display and set mode to alternate sweep.

   Figure A shows inverted video (top) and continuous subcarrier (bottom).

5. Adjust the oscilloscope for A plus B mode.

6. Adjust subcarrier phase and fine level at the generator or delay line for a null at burst as shown in Figure B.

7. Adjust the oscilloscope for chop mode, non-inverted video, and adjust vertical positions to exactly overlay subcarrier and sync.

8. Adjust the oscilloscope delayed sweep for a display showing the leading edge of sync and the subcarrier. A proper phase relationship requires coincidence at the 50% points of the leading edge of sync and the subcarrier zero crossings. See Figure C. An improper phase relationship is shown in Figure D.

Color Frame Pulse (V1) Identification

9. Adjust the SC/H phase as described in steps 1 through 8, for proper coincidence.

10. Trigger the oscilloscope on the leading edge of the V1 pulse with video and subcarrier connected to the two input channels. See Figure E.

11. Increase the oscilloscope sweep rate and, using the delayed sweep option, view a display showing the first leading edge of sync following the trigger.

12. If the negative transition of the subcarrier is coincident with the leading edge of sync, the triggering V1 pulse is a color frame identification pulse that occurs on line 11 of field 1. See Figure F.

NOTE: The SC/H phase is easiest to observe on a display that is horizontally triggered. Because of the low repetition rate of V1 and the fast sweep rates (50nS/div.) required, only the direction of subcarrier signal can be easily observed by triggering on V1.

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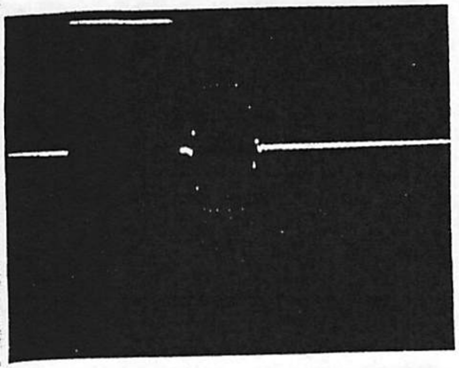
Fig. A

INVERTED VIDEO

CONTINUOUS SUBCARRIER

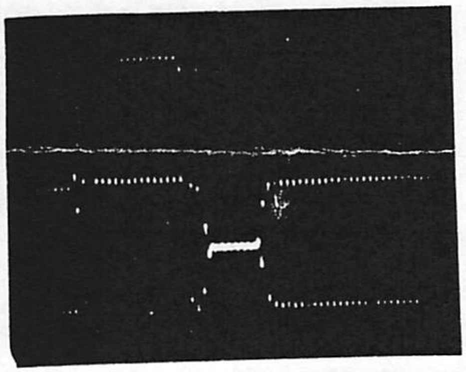
Fig. B

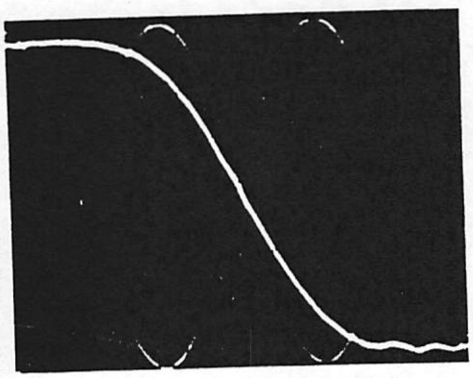
SUBCARRIER

Fig. C

CORRECT SC/H PHASE

COINCIDENCE AT 50% POINT OF LEADING EDGE OF SYNC AND SUB-CARRIER ZERO CROSSINGS.

(50nS/Div.)

LEADING EDGE OF SYNC

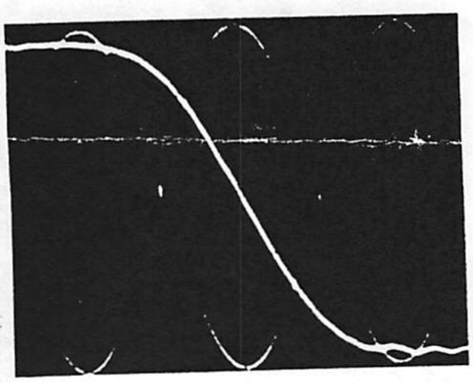
SUBCARRIER

Fig. D

INCORRECT SC/H PHASE

NON-COINCIDENCE AT 50% POINT OF LEADING EDGE OF SYNC AND SUB-CARRIER ZERO CROSSINGS.

(50nS/Div.)

LEADING EDGE OF SYNC

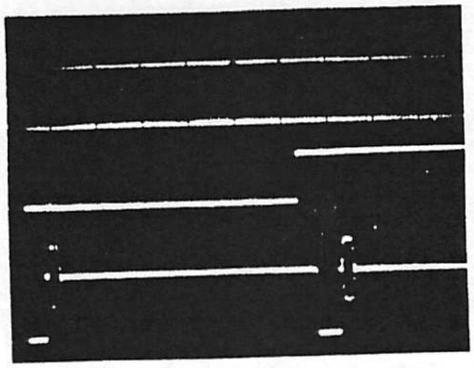
SUBCARRIER

Fig. E

V1 PULSE (TRIGGER VIEW)

VIDEO

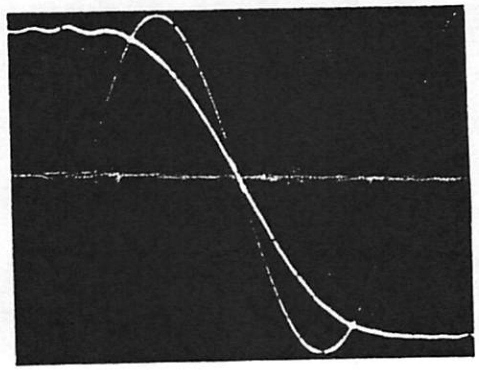
SUBCARRIER

Fig. F

COINCIDENCE

(50nS/Div.)

LEADING EDGE OF SYNC

Figures

A: Inverted Video and Continuous Subcarrier
B: Subcarrier Phase Adjusted for Null at Burst
C: Properly Phased SC/H Signal
D: Improperly Phased SC/H Signal (70 phase error)
E: Subcarrier, V1 Pulse, and Video Display
F: Leading Edge of Line 11 Field 1 and SC

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Video waveform monitoring

By Margaret Feisel

Waveform and vector monitors tell a story about video signal quality that a picture monitor can't.

Someday, you may be able to walk into a TV studio, flip a single switch and instantly send perfect TV signals throughout the facility. Today's studios, however, simply don't work that way. Equipment malfunctions, signals become distorted and a good deal of adjustment is required. Because of these problems, the waveform monitor and vectorscope are necessities at every video facility. Both instruments have existed for more than 20 years, but new models with new features are still being introduced. They continue to provide the best method of getting the necessary information to you. They let you *see* the signals.

Waveform and vector monitors are specialized oscilloscopes adapted for the video environment. The waveform monitor is much like a traditional oscilloscope, operating in a voltage vs. time mode. Its time base triggers automatically on sync pulses in the TV signal, producing line- and field-rate sweeps. Filters, clamps and other circuits process the video signal for specific monitoring needs.

The vectorscope operates in an X–Y voltage vs. voltage mode to display chrominance information. It decodes the signal in much the same way as a TV monitor or receiver to extract color information.

The two instruments serve separate, distinct purposes as they sit side by side in a rack, monitoring the same signal. Some newer models combine functions, packaging both monitor types on one chassis with a single CRT. Others have a communications link between two separate instruments.

Beyond basic signal monitoring, the instruments provide a means to identify and analyze signal aberrations. If the signal is distorted, these instruments allow a technician to learn the extent of the problem and to locate the offending equipment.

Basic waveform measurements

Waveform monitors are used to evaluate the amplitude and timing of video signals and to show timing relationships between two or more signals. The familiar color-bar pattern is the only signal required for these basic tests.

A word of caution about the color-bar signals is necessary to avoid confusion and possible inaccurate measurements. *All color bars are not created equal.* Some generators offer a choice of 75% or 100% amplitude bars. Although sync, burst and setup amplitudes remain the same for the two color-bar signals, peak-to-peak amplitudes of high-frequency chrominance information and low-frequency luminance levels change. The saturation of color, a function of chrominance and luminance amplitudes, remains constant at 100% in both modes. The 75% bar signal has 75% amplitude with 100% saturation. In 100% bars, amplitude and saturation are both 100%.

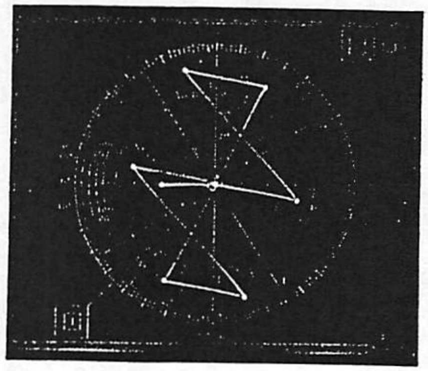

Figure 1. Color bars on a waveform monitor, 75% amplitude, 100IRE white, 2H sweep.

Chrominance amplitudes for 100% bars exceed the maximum amplitude that should be transmitted. Therefore, 75% amplitude color bars, with no chrominance information exceeding 100IRE, are the standard amplitude bars for NTSC. In the 75% mode, a choice of 100IRE or 75IRE white reference level may be offered. Figure 1 shows 75% amplitude bars with a 100IRE white level. Either white level can be used to set levels, but operators must be aware of which signal has been selected. SMPTE bars have a white level of 75IRE as well as a 100IRE white flag.

For a waveform monitor to make precise evaluations of signals, the monitor itself must be functioning properly. It should be periodically taken to a service center for calibration. The internal calibration signal for precise gain and sweep adjustments should be used regularly.

Select the calibration signal on the front panel and, if necessary, adjust the vertical gain calibration control until the square wave signal is exactly 140IRE units in amplitude. (Some instruments may require settings of 100IRE units. Consult the manual.)

If the signal is also a sweep calibrator, adjust the horizontal calibration control so that the square wave crosses the graticule base line at the major division marks. Figure 2 shows a typical calibrator signal. The adjustments are generally done with a screwdriver; don't confuse calibration controls with variable gain knobs.

Another preliminary consideration is the dc restorer setting. The dc restorer normally should be *on*, to stabilize the display from variations in average picture level (APL). Some instruments offer slow or fast dc restorer speeds. The slow speed maintains an average dc level, but permits the observation of low-frequency abnormalities, such as 60Hz hum. The fast speed removes most of the hum.

Most waveform monitors have filters that process the signal in order to display certain components. The *flat* response is best for basic monitor functions, because it displays the entire signal. In Figure 1, a flat response has been selected. The

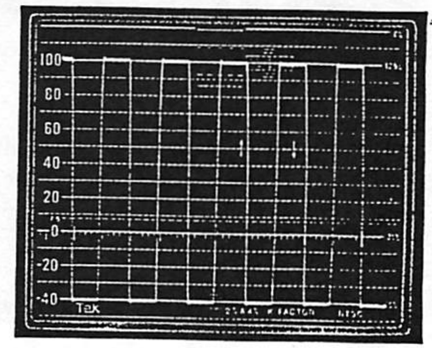

Figure 2. A waveform monitor calibrator signal.

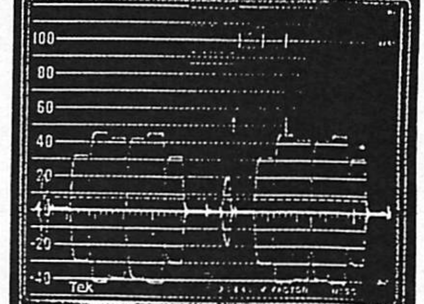

Figure 3. Chroma filter response with color bars, 2H sweep.

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a response filter removes luminance and displays only chrominance, as shown in Figure 3. The *low-pass* filter removes chrominance, leaving only low-frequency luminance levels in the display. (See Figure 4.)

The IRE filter was originally designed to average out high-level, fine-detail peaks on a monochrome TV signal, aiding the operator in setting brightness levels. The IRE response removes most, but not all, of the chrominance. Even so, the IRE mode can be useful for observing luminance levels, as shown in Figure 5.

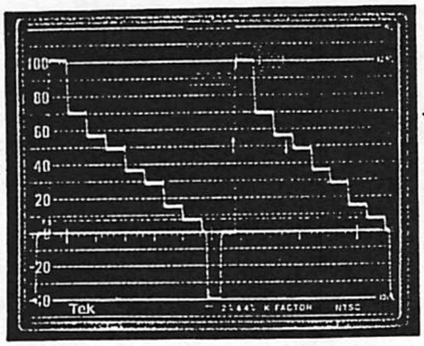

Figure 4. Low-pass filter response with color bars, 2H sweep.

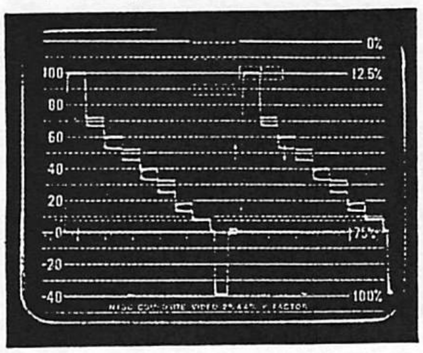

Figure 5. IRE filter response with color bars, 2H sweep.

Video amplitudes

The overall amplitude of the video signal is an important parameter to monitor. Deviations from the nominal 1V signal, expressed in IRE units or as percentages, are referred to as *insertion gain* or *loss*. Any equipment in the video path may change the gain, with the errors cascading to result in severe picture impairments. Even small changes in brightness can be perceived by the human eye.

The insertion gain errors, whether too large or too small a signal amplitude, may eventually manifest themselves as signal distortions. It is, therefore, important for each piece of equipment to accurately transfer a 1V signal at its input to a 1V signal at the output. Insertion gain is measured at the output of every active device in the signal path.

To check overall amplitude, position the displayed waveform vertically, until the blanking (back porch) level overlays the 0IRE graticule line. A vertical scale on the graticule divides the standard 1V video signal display into 140IRE units—100IRE above the base line and 40IRE below. The white level should come just to the 100IRE or 75IRE mark, depending on the color bars used.

Sync should extend to −40IRE, while setup (the black part of the color bars) should be at 7.5IRE. Setup and peak white are generally the only luminance levels checked with the color-bar signal. A linearity signal can be used to check intermediate luminance levels, but linearity is usually not considered necessary for basic insertion gain evaluation.

Some chrominance checks should be part of an insertion gain measurement procedure. Check (and adjust) the burst amplitude, making certain that it is 40IRE peak to peak, centered on the 0IRE mark. Verify that peaks of chrominance on the first and second 75% signal bars reach the 100IRE line. A vectorscope is better suited to precisely evaluate chrominance amplitudes and will be discussed later in this article.

Sync pulses

The duration and frequency of the TV system sync pulses also must be monitored. Horizontal sync width should be watched closely. Most waveform monitors have 0.5μs or 1μs per division magnification (MAG) modes, which help to verify that H-sync width is between 4.4μs and 5.1μs, when measured at the −4IRE point. On waveform monitors with good MAG registration, sync appearing in the middle of the screen in the 2-line mode remains centered when the sweep is magnified.

It is a good idea to check the rise and fall times of sync and the widths of the front porch and entire blanking intervals. Examine burst and count the cycles. There should be between eight and 11 cycles of subcarrier.

Check the vertical interval for correct format and measure the timing of the equalizing pulses and vertical sync pulses. The acceptable limits for these parameters are shown in the most recent FCC pulse width specification, reproduced in Figure 6.

System timing

The sync pulses of all signals must be in the same phase at the point in the studio at which they are combined. If they are not properly timed, the viewer will see a horizontal shift when the program switches from one source to another. Even though all signals are locked to the house reference, timing errors arise as signals travel through different cable lengths, delaying one with respect to the other. Therefore, timing delay for each piece of equipment must be adjusted to bring all signals into coincidence at the switcher.

To compare signal timing adjustments, connect a waveform monitor at the output of the switcher. Select the external reference with the house reference (probably blackburst) signal connected to the

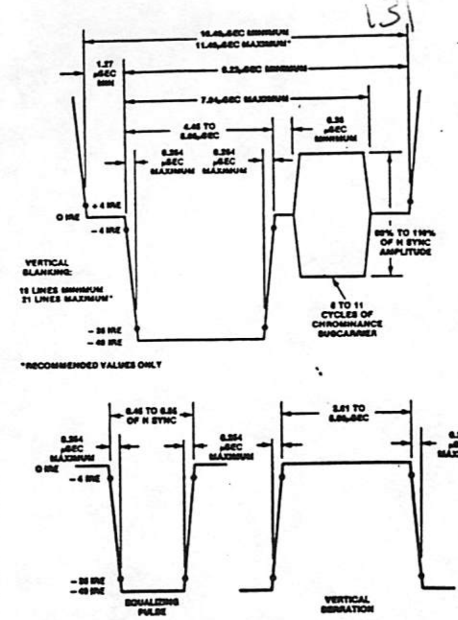

Figure 6. FCC pulse-width requirements as of March 14, 1985.

external reference input. First, select this reference signal on the switcher and display it on the waveform monitor in the *2H MAG* sweep mode. Adjust the horizontal position control on the waveform monitor to place the 50% amplitude point of the leading edge of sync on one of the horizontal axis graticule marks. *(Do not change this horizontal position knob setting until system timing is finished.)* Now, switch through the video sources one by one, adjusting each source so that the leading edge of sync falls on the same graticule mark as the reference signal.

Why time to the blackburst house reference? The truth is that black really does not need to be phased, if it is never to be a program input. Any signal could be brought up first and other signals matched to it. However, something has to be the reference and blackburst is a handy place to start.

Monitoring

The procedures described thus far are performed with test signals before anything goes on-air. During editing or broadcasting, set the waveform monitor in the 2H sweep mode. Keep an eye on it in case levels need adjustment. If something goes drastically wrong with the picture, check the waveform monitor first. Has sync or burst disappeared? Are the amplitudes correct? The waveform monitor will provide clues to the nature of the problem.

Some waveform monitors have a dual filter mode, enabling the video operator to observe luminance levels and overall amplitudes at the same time. In this mode, the instrument switches between the flat and low-pass filters. With a 2H sweep selected, the display on the left is low-pass filtered, while information to the right is unfiltered. The *line select*

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mode (which will be discussed in more detail) is another useful feature for monitoring live signals.

Basic vectorscope measurements

The vectorscope displays chrominance amplitudes, aids hue adjustments and simplifies matching relative burst phases of multiple signals. These functions require only the color-bar test signal.

To evaluate and adjust the chrominance in the TV signal, observe color bars on the vectorscope. The vectorscope should be in its calibrated gain position. Adjust the vectorscope phase control to place the burst vector at the 9 o'clock position, and note the vector dot positions with respect to the six boxes marked on the graticule. If everything is well adjusted, each dot will fall on the crosshairs of its corresponding box, as shown in Figure 7.

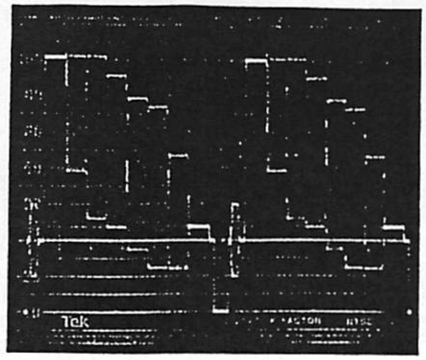

***Figure 7.** Color bars on a vectorscope.*

Chrominance amplitudes

The chrominance amplitude of a video signal determines the correct intensity or brightness of color. If the amplitudes are correct, each color dot falls on the crosshairs in the corresponding graticule box. If vectors overshoot the boxes, chrominance amplitude is too large; if they undershoot, it is too small.

The boxes at each color location can be used to quantify the error. In the radial direction, the small boxes indicate a ± 2.5IRE deviation from the standard amplitude. The large boxes indicate a $\pm 20\%$ error. Adjust chrominance amplitude on the signal source to get as close to zero error as possible.

Hue control

The hue control on TV equipment changes the phase of the color burst with respect to the rest of the signal. If it is improperly adjusted, viewers will not see true colors. Even small hue errors are undesirable, because very slight skin tone variations are noticeable. When the hue control is adjusted, the burst will remain in the 9 o'clock position as the other dots rotate around it, falling into their boxes. Again, the graticule boxes indicate the extent of the error in the circumferential direction. The small boxes represent $\pm 2.5°$ error, while the large boxes indicate $\pm 10°$.

Burst phasing

A vectorscope can be used to make sure that signals from various sources have the same phase of burst. If all burst signals are not properly phased, color shifts will occur when switching between sources. This adjustment is related to sync timing with a waveform monitor and is done in much the same way.

Connect the test instrument at the output of the switcher and select the external reference on the vectorscope. Switch up the reference signal first and use the vectorscope phase control to set the burst at 9 o'clock. (Do not move the phase control until burst phasing is finished.) Then, switch up each source, adjusting the phase control on the source until its burst vector is also at 9 o'clock.

Monitoring

Vectorscopes are primarily setup tools and are less useful than waveform monitors for watching live video. Burst can be distinguished, but the picture information is usually a blur. An exception to this is the use of a vectorscope to match the tints in backgrounds.

Other capabilities

The functions discussed thus far are

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the common uses for vector and waveform monitors and are routinely performed in most studios. There are additional uses, however, that require an instrument with a more advanced feature set, a more skilled operator or both. Some of these involve identifying problems, while others are precise measurements to quantify equipment aberrations.

Other test signals, including modulated staircase or multiburst, are required for many of these tests. It is important to take a good look at how these signals appear immediately after they come out of the generator. Knowing what the undistorted waveform looks like simplifies the identification of distortions.

Line select

Some models of waveform monitors and vectorscopes have line select capability. They can display only one or two lines out of the entire video frame of 525 lines. (In the normal display all of the lines are overlaid on top of one another.) The principal use of this single line feature is to monitor VITS signals. VITS allows in-service testing of the transmission system, because precise evaluation of signal quality is possible with test signals.

A full-field line selector generally drives a picture monitor output with an intensifying pulse. The pulse causes a single horizontal line on a picture monitor to be highlighted. This indicates where the line selector is within the frame and correlates the waveform monitor display with the picture.

Linear distortion: frequency response

Frequency response refers to the TV system's capability to respond uniformly to signal components of different frequencies and is generally evaluated with a waveform monitor. Different signals are required to check the various parts of the frequency spectrum. If the signals are all faithfully reproduced on the waveform monitor after passing through the video system, it is safe to assume that there are no serious frequency response problems.

At very low frequencies, look for externally introduced distortions, such as power-line hum or power-supply ripple, and distortions resulting from inadequacies in the video equipment itself. Low-frequency distortions will probably appear on the TV screen as flickering or slowly varying brightness. Low-frequency interference can be seen on a waveform monitor if the dc restorer is operating in the slow mode and a 2-field sweep is selected. Sine wave distortion from ac power-line hum is quite evident in Figure 8.

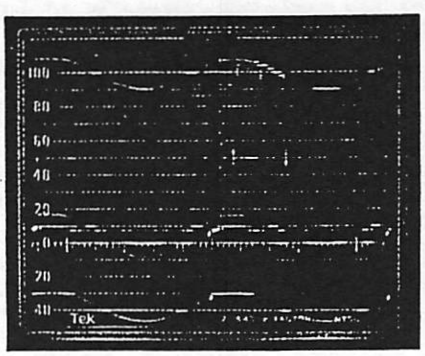

Figure 8. A 2-field sweep on a waveform monitor, showing ac mains hum distortion.

A bouncing APL signal can be used to detect distortion in the system itself. Vertical shifts of the blanking and sync levels indicate the possibility of low-frequency distortion.

Field-rate distortions appear as a difference in shading from the top to the bottom of the picture. A field-rate 60Hz square wave is best for measuring field-rate distortions. Distortion occurs as tilt in the waveform in 2-field mode with the dc restorer off. If a 60Hz square wave is not available, a window signal can also be used.

Line-rate distortions manifest themselves as streaking, shading or poor picture stability. To detect errors of this nature, look for tilt in the bar portion of a pulse-and-bar signal. The waveform monitor should be in the 1H or 2H mode with the fast dc restorer selected for the measurement.

The multiburst signal is used to test the high-frequency response of a system. The multiburst, shown in Figure 9, includes packets of discrete frequencies within the TV passband, with the higher frequencies toward the right of each line. The highest frequency packet is at about 4.2MHz, which is the upper frequency limit of the system. The next packet to the left is near the color subcarrier frequency (3.58MHz) and is used to check the chrominance transfer characteristics.

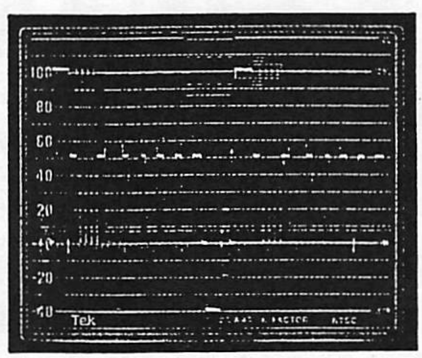

Figure 9. Undistorted multiburst, 2H sweep.

The remaining packets are distributed down to 500kHz. The most common distortion is high-frequency rolloff, seen on the waveform monitor as reduced amplitude packets for the higher frequencies (see Figure 10). The TV picture exhibits loss of fine detail and color intensity when this type of distortion is present. High-frequency peaking, appearing on the waveform as higher amplitude packets at the higher frequencies, causes

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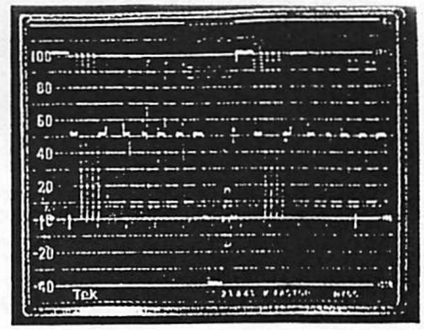

Figure 10. Multiburst signal exhibiting high-frequency rolloff.

ghosting on the picture.

Non-linear distortion: differential phase

Non-linear, or level-dependent, distortions are another type of picture impairment. One non-linear distortion is differential phase (dΘ), which is present if a change in luminance level produces a change in the chrominance phase. If the distortion is severe enough, TV viewers will note that the hue of an object changes as its brightness changes. A modulated staircase or ramp is used to measure this. Either signal places chrominance of uniform amplitude and phase at different luminance levels. Figure 11 shows a 100IRE modulated ramp. Because dΘ may change with changes in APL, several measurements of this distortion are necessary to fully evaluate system response. Measure dΘ at the center and at the two extremes of the APL range available on the test signal generator.

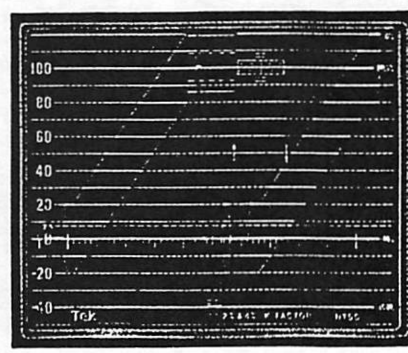

Figure 11. A 100IRE modulated ramp, 2H sweep.

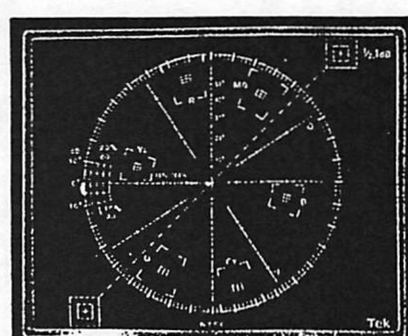

Figure 12. Differential phase of 5° on vectorscope.

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To measure dΘ on a vectorscope, increase the gain control until the vector dot is on the edge of the graticule circle. Use the phase shifter to set the vector to the 9 o'clock position. Phase error will appear as circumferential elongation of the dot. The vectorscope graticule has a scale marked with degrees of dΘ error. Figure 12 shows a dΘ error of 5°.

More information can be obtained from a swept R−Y display, which is a feature of waveform monitor and vectorscope systems. If one or two lines of demodulated video from the vectorscope are displayed on a waveform monitor, differential phase appears as tilt across the line. In this mode, the phase control should be adjusted to place the demodulated video on the base line, which is equivalent in phase to the 9 o'clock position on the vectorscope. Figure 13 shows a dΘ error of 5°; the amount of tilt is measured against a vertical scale.

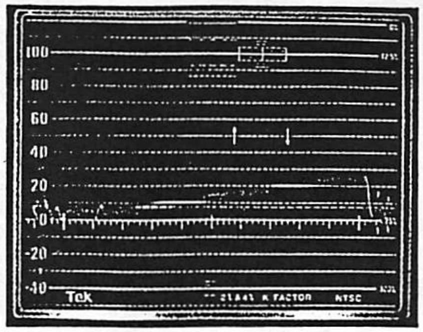

Figure 13. Differential phase of 5° on waveform monitor.

This mode is useful in troubleshooting. By noting where along the line the tilt begins, it is possible to figure out at what dc level the problem starts to manifest itself. In addition, field-rate sweeps enable the operator to look at dΘ over the field.

A variation of the swept R−Y display may be available in some instruments for precise measurement of differential phase. Highly accurate measurements can be made with a vectorscope that has a precision phase shifter and a double-trace mode. This method involves nulling the lowest part of the waveform with the phase shifter, then using a separate calibrated phase control to null the high end of the waveform. A readout in tenths of a degree is possible.

Non-linear distortion: differential gain

Differential gain (dG), another non-linear distortion, refers to a change in chrominance amplitude with changes in luminance level. That is, the vividness of a colored object will change when the brightness of the scene changes. The modulated ramp or modulated staircase is also used to evaluate this impairment. Again, make the measurement on signals with different APL levels.

To measure differential gain with a vectorscope, set the vector to the 9 o'clock position and use the variable gain to bring it to the edge of the graticule circle. Differential gain error appears as a lengthening of the vector dot in the radial direction. The dG scale at the left side of the graticule can be used to quantify the error. Figure 14 shows a dG error of 10%.

Differential gain can also be evaluated on a waveform monitor by using the chroma filter and examining the amplitude of the chrominance from a modulated staircase or ramp. Put the waveform monitor in 1H sweep and use the variable gain to set the amplitude of the chrominance to 100IRE. If the

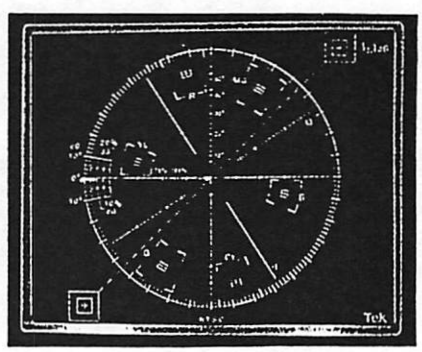

Figure 14. Differential gain of 10% on a vectorscope.

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chrominance amplitude is not uniform across the line, there is a dG error. The reduced chrominance amplitude at the right side of Figure 15 is an example of dG distortion. With the gain normalized

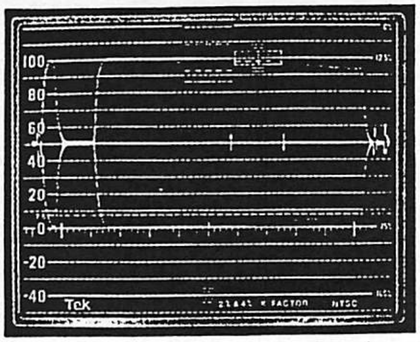

Figure 15. Differential gain of 10% on waveform monitor.

to 100IRE, the error can be expressed as a percentage.

Finally, dG can be precisely evaluated with a swept display of demodulated video. This is similar to the single trace R–Y methods for differential phase. The B–Y signal is examined for tilt when the phase is set so that the B–Y signal is at its maximum amplitude. The tilt can be quantified against a vertical scale.

K-factor

The lines and small boxes near the top of the waveform monitor graticule are K-factor or quality constant scales. The K-factor system is another means to quantify signal degradation. A series of subjective viewer reaction tests generated data that correlated the relative degradation of the picture to a measured amount of all the distortions observed on a sine² pulse-bar signal. From the results of those tests, the K-factor established quality standards for TV signals from slight to severe picture degradation.

The K-factor markings line up horizontally with the pulse-bar portion of the FCC composite signal when the waveform monitor is set for a 1H sweep. The dashes and solid lines on the graticule represent ±2% and ±4% K-factor, respectively. A 5% K-factor distortion is said to be detectable by skilled observers, while 3% is not noticeable.

ICPM

Modern TV receivers use a method known as intercarrier sound to reproduce audio information. Sound is recovered by beating the audio carrier against the video carrier, producing a 4.5MHz IF signal, which is, in turn, demodulated to produce the sound. From the interaction between the audio and video portions of the signal, certain distortions in the video at the transmitter can produce audio buzz at the receiver. Distortions of this type are referred to as *incidental carrier phase modulation* or *ICPM.*

The advent of stereo audio for television has increased the importance of measuring this parameter at the transmitter, because the buzz is more objectionable in stereo broadcasts. It is generally suggested that less than 3° of ICPM be present.

ICPM is measured with a high-quality demodulator with a synchronous detection mode and an oscilloscope, operated in a high-gain X–Y mode. Some waveform and vector monitors have such a mode. Video from the demodulator is fed into the Y input of the scope and quadrature out is fed to the X input terminal. Low-pass filters make the display easier to resolve.

An unmodulated 5-step staircase signal produces a polar display, which is shown in Figure 16, on a special graticule developed for this purpose. Note that the bumps all rest in a straight vertical line, if there is no ICPM in the system. Tilt indicates an error, as shown in Figure 17. The graticule is calibrated in degrees per radial division for differential gain settings. Adjustment, although not measurement for the error, can be done without a graticule.

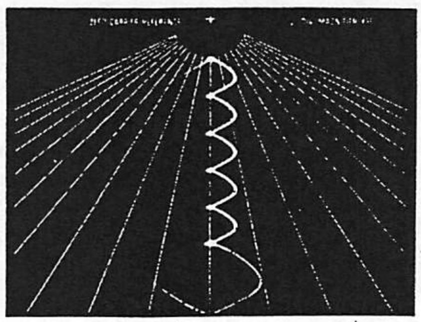

Figure 16. A waveform display showing no ICPM distortion.

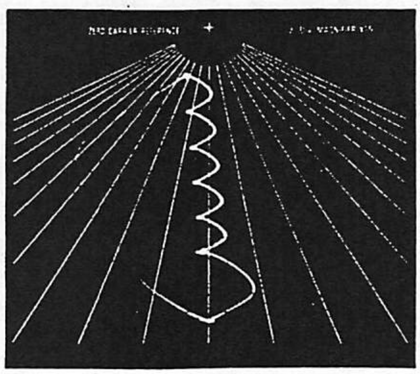

Figure 17. A waveform display showing 5° ICPM distortion.

And so on...

This discussion of capabilities of waveform and vector monitors is by no means all-inclusive. Special test signals and measurement techniques have been developed to evaluate chrominance-to-luminance delay, group delay and many other parameters. Some instruments have the capability to look at SC/H phase. Instructions for performing these measurements and more detailed information on techniques already discussed can usually be found in your monitoring equipment manuals.

An effective test plan

Waveform and vector monitors are useful instruments, but the effectiveness of test and measurement procedures depends heavily upon the people who operate them. Someone must decide which parameters need to be evaluated and what the acceptable limits are for each one. Someone must select the test equipment initially. And finally, someone must be responsible for making those measurements on a routine basis.

Whether these functions are performed by one person or by members of a large staff, their judgment is important. There are no hard and fast rules, no pat answers. Technical people involved in television are always trying to get the best picture for the viewer while staying within a budget and meeting FCC requirements. In spite of this common goal, each facility will have different test and measurement requirements.

The first concern, and probably the most difficult one, is deciding what to measure and how much error to tolerate. Most facilities will want to make the checks discussed in this article, but it is impossible to generalize much further. Deciding on acceptable limits is not easy. Is 5° of dG too much? It may be perfectly acceptable for a transmitter site, while a dG of 1° may be objectionable in the studio. The operators of each facility must decide for themselves and it is, therefore, a good idea to learn how various impairments affect the picture.

The choice of test and measurement equipment is usually based on the feature set required, the cost, and personal preferences of the operators. When only a few models were available, a clear line could be drawn between monitoring equipment and measurement equipment. Although some instruments are still clearly one or the other, that line has blurred considerably in recent years. Many newer models fall in between, making it necessary to analyze the instruments feature by feature to find the best fit. Keep in mind the experience and technical skills of those who will operate the equipment most frequently, and try to find an appropriate user interface.

Finally, make sure that all of the checks and setup procedures are performed regularly by a competent operator. Good equipment and a well-defined system checkout procedure will be effective only if they are used. The person performing the check must be able to recognize when something is wrong, even if someone else must be called to fix the problem. A good waveform monitor and vectorscope, when combined with some basic skills on the part of the operator, can help achieve the goal of delivering consistently high-quality video.

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GLOSSARY OF TELEVISION TERMS

AC-COUPLED A connection which removes the constant voltage (DC component) on which the signal (AC component) is riding. Usually implemented by passing the signal through a capacitor.

AM Amplitude Modulation (AM) is the process by which the amplitude of a high-frequency carrier is varied in proportion to the signal of interest. In the NTSC television system, AM is used to encode the color information and to transmit the picture.

Several different forms of AM are differentiated by filtering of the sidebands and whether or not the carrier is suppressed. Double sideband suppressed carrier is used to encode the NTSC color information, while the signal is transmitted with a vestigial sideband scheme.

APL Average Picture Level. The average signal level (with respect to blanking) during active picture time, expressed as a percentage of the difference between the blanking and reference white levels.

BACK PORCH The portion of the video signal which lies between the trailing edge of the horizontal sync pulse and the start of the active picture time. Burst is located on back porch.

BANDWIDTH The range of frequencies over which signal amplitude remains constant (within some limit) as it is passed through a system.

BASEBAND Refers to the composite video signal as it exists before modulating the picture carrier. Composite video distributed throughout a studio and used for recording is at baseband.

BLACK BURST Also called "color black", black burst is a composite video signal consisting of all horizontal and vertical synchronization information, burst, and usually setup. Typically used as the house reference synchronization signal in television facilities.

BLANKING LEVEL Refers to the 0 IRE level which exists before and after horizontal sync and during the vertical interval.

BREEZEWAY The portion of the video signal which lies between the trailing edge of the horizontal sync pulse and the start of burst. Breezeway is part of back porch.

BROAD PULSES Another name for the vertical synchronizing pulses in the center of the vertical interval. These pulses are long enough to be distinguished from all others, and are the part of the signal actually detected by vertical sync separators.

BURST A small reference packet of the subcarrier sine wave, typically 8 or 9 cycles, which is sent on every line of video. Since the carrier is suppressed, this phase and frequency reference is required for synchronous demodulation of the color information in the receiver.

B-Y One of the color difference signals used in the NTSC system, obtained by subtracting luminance from the blue camera signal. This is the signal which drives the horizontal axis of a vectorscope.

CHROMINANCE Chrominance refers to the color information in a television picture. Chrominance can be further broken down into two properties of color: hue and saturation.

CHROMINANCE SIGNAL The high-frequency portion of the video signal which is obtained by quadrature amplitude modulation of a 3.58 MHz subcarrier by R-Y and B-Y.

COLOR DIFFERENCE SIGNALS Signals used by color television systems to convey color information in such a way that the signals go to zero when there is no color in the picture. R-Y, B-Y, I and Q are all color difference signals.

COMPONENT VIDEO Video which exists in the form of three separate signals, all of which are required in order to completely specify the color picture. For example: R, G and B or Y, R-Y, and B-Y.

COMPOSITE VIDEO A single video signal containing all of the necessary information to reproduce a color picture. Created by adding quadrature amplitude modulated R-Y and B-Y to the luminance signal.

CW Continuous Wave. Refers to a separate subcarrier sine wave used for synchronization of chrominance information.

dB (DECIBEL) A decibel is a logarithmic unit used to describe signal ratios. For voltages,

$$dB = 20 \, Log_{10} \left(\frac{V_1}{V_2} \right)$$

DC-COUPLED A connection configured so that both the signal (AC component) and the constant voltage on which it is riding (DC component) are passed through.

DC RESTORER A circuit used in picture monitors and waveform monitors to clamp one point of the waveform to a fixed DC level.

DEMODULATOR In general, this term refers to any device which recovers the original signal after it has modulated a high frequency carrier. In television, it may refer to:

(1) An instrument, such as a Tektronix 1450, which takes video in its transmitted form (modulated picture carrier) and converts it to baseband.

(2) The circuits which recover R-Y and B-Y from the composite signal.

EQUALIZER The pulses which occur before and after the broad pulses in the vertical interval.

ENVELOPE DETECTION A-demodulation process in which the shape of the RF envelope is sensed. This is the process used by a diode detector.

FIELD In interlaced scan systems, the information for one picture is divided up into two fields. Each field contains one half of the lines required to produce the entire picture. Adjacent lines in the picture are in alternate fields.

FM Frequency Modulation (FM) is the process by which the frequency of a carrier signal is varied in proportion to the signal of interest. In the NTSC television system, audio information is transmitted using FM.

FRAME A frame contains all the information required for a complete picture. For interlaced scan systems, there are two fields in a frame.

FRONT PORCH The portion of the video signal between the end of active picture time and the leading edge of horizontal sync.

GAMMA Since picture monitors have a nonlinear relationship between the input voltage and brightness, the signal must be correspondingly predistorted. Gamma correction is always done at the source (camera) in television systems: the R, G and B signals are converted to $R^{1/\gamma}$, $G^{1/\gamma}$ and $B^{1/\gamma}$. Values of about 2.2 are typically used for gamma.

GENLOCK The process of locking both sync and burst of one signal to sync and burst of another, making the two signals completely synchronous.

GRATICULE The scale which is used to quantify the information on a waveform monitor or vectorscope display. Graticules may either be screened onto the faceplate of the CRT itself (internal graticule), or onto a piece of glass or plastic which fits in front of the CRT (external graticule). They can also be electronically generated.

HARMONIC DISTORTION If a sine wave of a single frequency is put into a system, and harmonic content at multiples of that frequency appears at the output, there is harmonic distortion present in the system. Harmonic distortion is caused by nonlinearities in the system.

HORIZONTAL BLANKING Horizontal blanking is the entire time between the end of the active picture time of one line and the beginning of active picture time of the next line. It extends from the start of front porch to the end of back porch.

HORIZONTAL SYNC Horizontal sync is the −40 IRE pulse occurring at the beginning of each line. This pulse signals the picture monitor to go back to the left side of the screen and trace another horizontal line of picture information.

HUE Hue is the property of color which allows us to distinguish between colors such as red, yellow, purple, etc.

HUM Undesirable coupling of the 60 Hz power sine wave into other electrical signals.

INTERCARRIER SOUND A method used to recover audio information in the NTSC system. Sound is separated from video by beating the sound carrier against the video carrier, producing a 4.5 MHz IF which contains the sound information.

IRE A unit equal to 1/140 of the peak-to-peak amplitude of the video signal, which is typically one volt. The 0 IRE point is at blanking level, with sync tip at −40 IRE and white extending to +100 IRE. IRE stands for Institute of Radio Engineers, the organization which defined the unit.

LINEAR DISTORTION Refers to distortions which are independent of signal amplitude.

LUMINANCE The signal which represents brightness, or the amount of light in the picture. This is the only signal required for black and white pictures, and for color systems it is obtained as a weighted sum ($Y = 0.3R + 0.59G + 0.11B$) of the R, G and B signals.

GLOSSARY OF TELEVISION TERMS

MODULATED When referring to television test signals, this term implies that chrominance information is present. (For example, a modulated staircase has subcarrier on each step.)

MODULATION A process which allows signal information to be moved to other frequencies in order to facilitate transmission or frequency-domain multiplexing. See AM and FM for details.

NONLINEAR DISTORTION Refers to distortions which are amplitude-dependent.

NTSC National Television System Committee. The organization which developed the television standard currently in use in the United States, Canada and Japan. Now generally used to refer to that standard.

PAL Phase Alternate Line. Refers to the television system used in Europe and many other parts of the world. The phase of the chrominance signal alternates from line to line to help cancel out phase errors.

QUADRATURE AM A process which allows two different signals to modulate a single carrier frequency. The two signals of interest Amplitude Modulate carrier signals which are the same frequency but differ in phase by 90 degrees (hence the Quadrature notation). The two resultant signals can be added together, and both signals recovered at the other end, if they are also demodulated 90 degrees apart.

QUADRATURE DISTORTION Distortion resulting from the asymmetry of sidebands used in vestigial sideband television transmission. Quadrature distortion appears when envelope detection is used, but can be eliminated by using a synchronous demodulator.

RF Radio Frequency. In television applications, RF generally refers to the television signal after the picture carrier modulation process.

RGB Red, Green and Blue. The three primary colors used in color television's additive color reproduction system. These are the three color signals generated by the camera and used by the picture monitor to produce a picture.

R-Y One of the color difference signals used in the NTSC system, obtained by subtracting luminance from the red camera signal. The R-Y signal drives the vertical axis of a vectorscope.

SATURATION The property of color which relates to the amount of white light in the color. Highly saturated colors are vivid, while less saturated colors appear pastel. For example, red is highly saturated, while pink is the same hue but much less saturated.

SETUP In NTSC systems, video black is typically 7.5 IRE above the blanking level. This 7.5 IRE level is referred to as the black setup level, or simply as setup.

SUBCARRIER The modulation sidebands of the color subcarrier contain the R-Y and B-Y information. For NTSC, subcarrier frequency is 3.579545 MHz.

SYNCHRONOUS DETECTION A demodulation process in which the original signal is recovered by multiplying the modulated signal with the output of a synchronous oscillator locked to the carrier.

TERMINATION In order to accurately send a signal through a transmission line, there must be an impedance at the end which matches the impedance of the source and of the line itself. Amplitude errors and reflections will otherwise result. Video is a 75 Ohm system, so a 75 Ohm terminator must be put at the end of the signal path.

UNMODULATED When used to describe television test signals, this term refers to pulses and pedestals which do not have high-frequency chrominance information added to them.

VECTORSCOPE A specialized oscilloscope which demodulates the video signal and presents a display of R-Y versus B-Y. The angle and magnitude of the displayed vectors are respectively related to hue and saturation.

VERTICAL INTERVAL The synchronizing information which appears between fields and signals the picture monitor to go back to the top of the screen to begin another vertical scan.

WAVEFORM MONITOR A specialized oscilloscope for evaluating television signals.

Y Abbreviation for luminance.

ZERO CARRIER REFERENCE A 120 IRE pulse in the vertical interval which is produced by the demodulator to provide a reference for evaluating depth of modulation.

Video Tape Recorders

The Invention of Magnetic Tape
Electronic editing, as it is now known, came about with the invention of recordable magnetic tape and the decks that play them back. Prior to this invention, video signals could not be stored so there was nothing to edit! All video would have to be captured on film and then edited that way. In that case, why not just shoot on film?

The Video Tape Recorder
The video tape recorder, like an audio tape recorder but far more complex, was invented by Ampex in 1955. The complexity of a video tape recorder is easily attributed to one fact: the speed and complexity of a video signal. The rate of changing electrical values is very high in a video signal.

There are 525 lines per frame of NTSC video. There are approximately 30 frames per second.

525 lines/frame x 30 frames/second = 15,750 lines every second. Each line represents changes in brightness from left to right of an image. Every change in brightness is some change in the electrical signal. When the line is bright, the electrical signal is high, and as the line gets progressively darker, the electrical signal becomes lower. A video tape recorder has to be able to record all the fluctuations in brightness of every line 15,750 times in *one second*. This is a lot of information to record. Consider how small a video tape is. At most, a video tape is two inches wide. Somehow, a video tape recorder can store more than 15,750 alterations of information on magnetic particles on a video tape in a space two inches wide. Of course, the tape is moving. If the tape moves at ten inches per second, you have a space of 10 inches x 2 inches wide = 20 inches total. Still, imagine recording over 15,750 changes of information in a space 20 inches total. The changes would be so small that your eye might not be able to distinguish them. This is why you need a microscope to see the tiny tracks on a video tape.

The math in the previous paragraph demonstrates a black and white NTSC video signal on a 2 inch video tape. Newer formats are 1" wide, 3/4", 1/2", 8mm and now even 6mm. The same amount of information is somehow recorded in even smaller spaces! A lot of information in a very small space is the name of the game.

What does it mean?
All of that information fit into such a tiny space means that the physical devices which read and write the information must be equally small, sensitive, and precise. If the devices reading the magnetic information off of a tape are even less than a millimeter off during their reading, they will read the wrong track (the tracks of video being 0.13mm wide on a 1" video tape). These measurements are so small they are hard to comprehend in every day life, but this is the reality of recording a video signal. Realizing the precision and size of the signals you are working with helps to appreciate what can and can't be done when you are editing.

Devices that Record and Play Video
A video tape recorder (VTR) has one primary purpose: to store the electrical video signal onto magnetic tape and to recreate an electrical video signal from the magnetic information on the tape. Opening a video tape recorder reveals that there are a lot of small components - little video systems within the VTR system - to achieve this goal. The most critical elements that all VTRs have are:

-a video signal input connector (BNC, RCA or UHF)
-a video signal output connector (BNC, RCA, or UHF)
-a record head (transducer)
-a playback head (transducer)
-a motor which winds the tape past the record and playback heads

The input connector gets the video signal into the video deck. Once in the VTR, it is processed and sent to the record head.

Another Transducer - the Record Head
A record head is simply an electricity to magnetism transducer. It takes electrical energy and transduces it to magnetic energy. This is based on the fact that electricity and magnetism are inherently related. One law of physics states that electricity running through a wire produces a proportional magnetic field. If the electrical signal is strong (a high voltage and current), the magnetic field produces around the wire is proportionally high. If the electrical signal running through a wire is weak, with a low voltage, the proportional magnetic field is weak.

A record head, then, is basically a tiny wire placed near the surface of the magnetic tape. When the electricity from the video input of the deck is high (because the picture is bright, for example), there will be a strong magnetic field emanated from the record head. A high number of magnetically sensitive particles will be polarized during that moment.

The Playback Head
The playback head is based on the inverse of the same fundamental: a magnetic field, placed near a wire, causes a proportional electrical voltage and current in a wire.

When the pre-recorded tape moves by the playback head (basically another tiny wire), the magnetic field emanated by the magnetic particles on the tape cause a current to be produced in the head which is proportional to the strength of the magnetic field. If a particular place on a video tape has a high number of polarized (magnetized in a specific direction) magnetic particles, there will be a proportionally high electrical signal produced in the playback head. If there is a particular place on a video tape with a low number of polarized magnetic particles, the result will be a low electrical current and voltage produced in the playback head. These electrical signals are sent from the playback head to the video output of the deck.

Both playback and record heads simply transduce magnetic energy to electrical or vice-versa. The invention of the magnetic tape recorder made possible storing any electrical information for future playback.

What's the big deal about video tape?
Without magnetic tape, all electrical signals only exist for one moment and are gone. This is live TV, live radio broadcast, live amplified musical performances. In these instances, electricity helps to enhance or broadcast information but it stops there.

Being able to reproduce electrical signals repeatedly is synonymous with recording oral stories and traditions with written language for future reference. It makes possible recording an event in time for future reference. With audio signals, it means the sounds of a president giving a speech or a grandmother telling family stories can be transduced to electrical information and then stored as magnetic information. Also, the light from an event like a parade, a tree growing, a child's first steps can be transduced to electricity and that information can be stored as magnetic information.

It also means that video information can be selectively chosen and transferred from one video tape to another. This was possible for years with film, but at last it was possible with the electrical signal. A new age of post-production was born!

recording engineers, but also by musicians who describe the sounds of synthesizers and other instruments in terms of waveforms.

Since different sound sources vibrate 'imperfectly,' waveforms other than the sine wave exist, and have a different sound char-

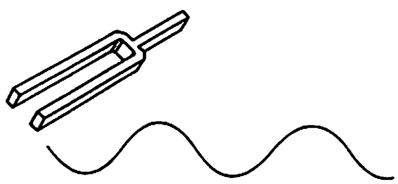


Figure 1.6 If we could plot the waveform of a tuning fork, it would resemble a sine wave.

acter from the 'pure' sine wave. There are many reasons why sources have distinct vibration characteristics. If the source is an instrument, everything from its construction material, to its size, to the way it's played contributes to the waveform.

Aside from the sine wave, some other mathematical waveforms are shown in Figure 1.7.

Those of you with synthesizers may in fact have some of these waveforms within your instrument. If so, and if you're unfamiliar with these sounds, call them up on the synthesizer and compare their timbre. While a sine waveform has a pure, flute-like tone, a square wave has a 'warmer,' reed-like tone, such as that of a clarinet. Triangle and sawtooth waves are more complex, and have 'harsher' tones, such as that of a saxophone. But the *primary*, or *fundamental*, waveform of a sound is only one component of a sound's overall, or *composite* waveform.

A composite waveform can become extremely complex. *Overtones*, or *harmonics*

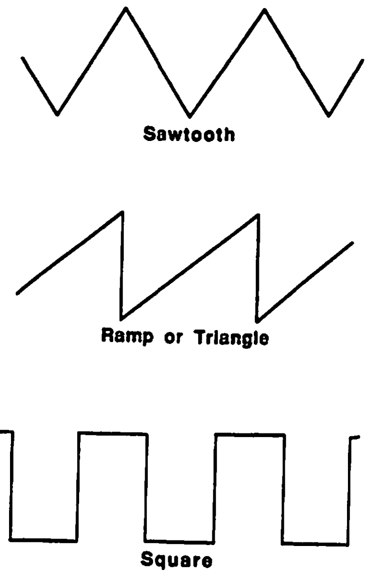


Figure 1.7 Some other simple waveforms, as are commonly found onboard many analog synthesizers.

(such as all the extra tones that can be heard when a gong is struck), serve to define an individual waveform. If two different instruments are played together at the same time, our ears sum together their waveforms.

In this manner, the music of an entire orchestra could be represented as one highly complex waveform at any given time. Figure 1.8 shows a diagram of a complex waveform. It's a single piano note, and it's represented three-dimensionally to show all the overtones.

Up to now, we've been talking about waveforms as sounds travelling through air.

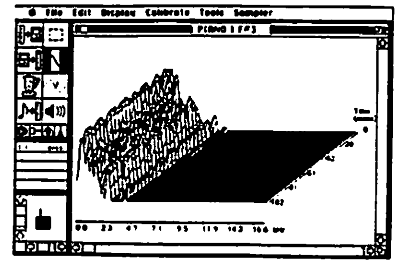


Figure 1.8 Digidesign's Apple Macintosh-based Sound Designer program analyzes the waveform of sound samples. This shows a single piano note, with all of its overtones. The number, relative amplitude, and pitch of the overtones over time all combine to help shape a sound's timbre.

When we record and play back sound, though, we're dealing with electrical waveforms. The three-dimensional waveform shown in Figure 1.8 has been generated by a computer, and it's electronic representation of an acoustic waveform. We'll learn more about the relationship between sound and electricity in the coming pages, and we'll learn more about how we hear sound starting on page 48.

* * * *

At this point, we now have an understanding of the basics of sound. While most recording engineers seldom refer to sound in terms of vibration and pressure, it's useful to be aware of how sound is created, how it travels, and how we hear it.

Many of the other terms we've covered, such as frequency response, the decibel, amplitude, and more, will be used throughout this book. Our basic understanding of what these mean and why they're used will be a good foundation for the next chapter, when we look at how recording works. □

CHAPTER 2: MAGNETIC RECORDING

In this chapter, we're going to learn some of the basic theory which makes multi-track recording possible. This includes:

- How sound is converted to electricity.
- How the electrical signal is stored on magnetic tape.
- Playing back the recorded signal.

We'll also take a look at:

- The transducer.
- Some fundamentals of electricity.
- The early development of multi-track recording.

Let's examine what most of us who make our own recorded music take for granted—the fascinating process of magnetic recording.

Transducers: Converting Sound To Electricity

Take a thin sheet of paper, hold it tightly with two hands up in front of your mouth, and talk into it. (Sure, it may seem a little weird, but this is in the name of *science!*) What do your fingers feel?

If you've talked loud enough, you should have felt the paper vibrating with your speech. Now, sing a single note, "aahhh"-style. The paper should vibrate more consistently than when talking. In fact, if you recall what we learned in Chapter One, you'll realize that the paper is vibrating at the same rate—at the same fundamental *frequency*—as the note you're singing.

What's the point of this exercise? To see how sound can be converted to a physical vibration—this is a necessary step in the conversion of sound to electricity.

It's this process that Leon Scott de Martinville observed in 1857, when he invented the precursor of Thomas Edison's phonograph. De Martinville used a scribe, attached to a sound-sensitive diaphragm, to etch sound waveforms onto a cylinder coated with lamp black. It wasn't until 1877 that Edison created a device that was able to read back recorded waveforms (on a wax-coated cylinder), and reconvert them to sound.

Both de Martinville's and Edison's inventions created mechanical representations of sound. In the modern recording world,

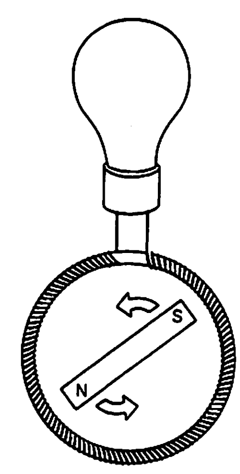


Figure 2.1 Electromagnetism.

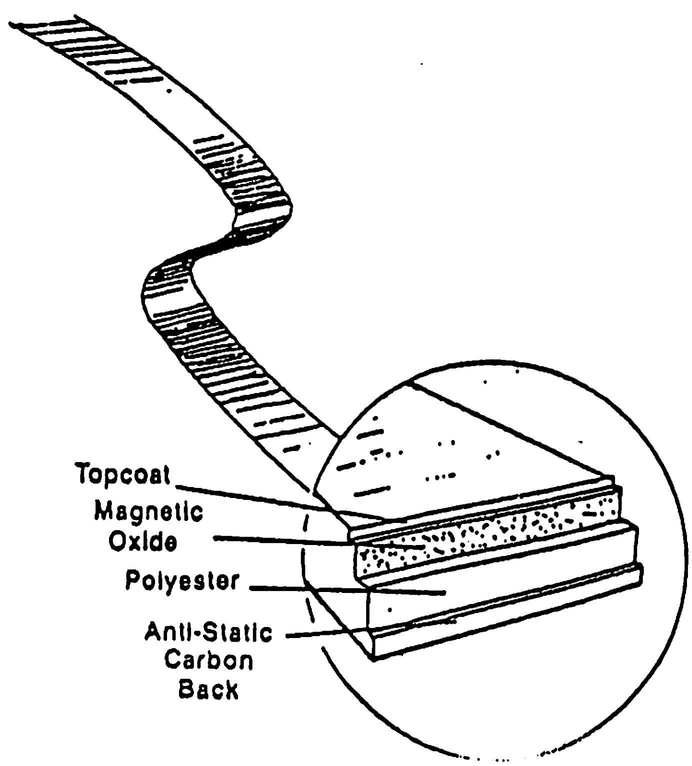
WHAT IS VIDEOTAPE?

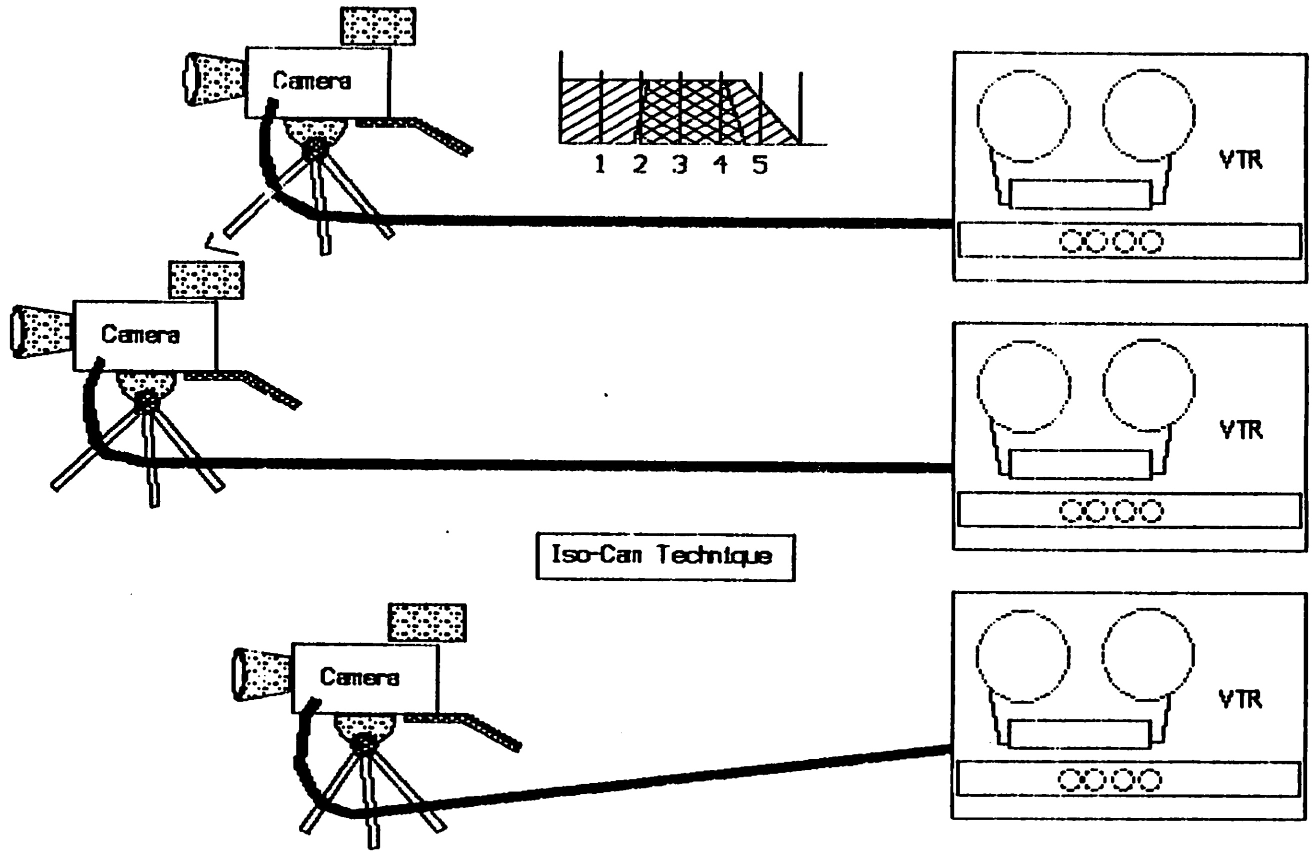
Film consists of various layers of chemicals on an acetate base. In contrast, videotape is a layered configuration whose surface has a magnetic coating of needle-shaped iron-oxide or chromium-dioxide particles which lie in a length-wise direction on the tape. These particles are bound to a polyester base. A recording head produces a magnetic field which stimulates the magnetically sensitive material on the videotape in such a way as to record and store picture and sound information which can be played back later.

Video is recorded and played back at 29.97 frames per second. With film, individual frames are visible. With videotape, individual frames are not visible to the eye, but are created electronically from the information stored on the videotape, using a scanning pattern.

DROPOUTS occur when a piece of the tape's magnetic oxide coating flakes off or is rough. This causes a "hole" or missing line of information in the picture. Dropouts appear as horizontal lines that dart through the image. They come in various sizes and durations, depending on the extent of the damage to the tape. Whenever tape is in contact with the heads, such as in playback or shuttling (same as the "search" function on home VCRs), it risks dropout.

THE MORE A TAPE IS RUN OVER THE HEADS, THE MORE RISK OF DROPOUT.





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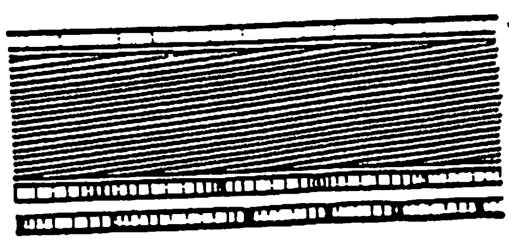
TRACKS

There are four signals recorded on a videotape during normal video recording:

1. Control Track Pulses (Sync Pulses)
2. Video
3. Audio channel 1
4. Audio channel 2

Each of these signals occupies a portion of the videotape called a "track". Each signal is recorded onto its track by a device called a "head". Each signal can be recorded separately.



5. Address Track: Some video tape recorders can record additional information on another track called the address track. This is where Time Code information is recorded (see CONTROL TRACK AND TIME CODE section) With Hi-8 and Betacam, the address track is separate and can be accessed independently from other tracks.

With S-VHS and VHS, Time Code is recorded either within the video track or on audio channel 2. The 3/4" format's address track is buried under the control track. Address track information must be recorded with the video and control track signals in the field, with these formats, or generated later on audio channel 2. There are two types of address track, longitudinal (most common) and vertical interval. A VTR which records and reads one may not be able to read or record the other.

Video Formats

To list all of the video formats ever made would require going through every painful detail of video history. Some formats never made it past the laboratories, others were released and suddenly turned to flops. Several have lasted consistently over the years.

The term video format does not necessarily imply a tape based system, though it usually does. The NTSC video signal, for example, is truly a broadcast signal format. It is used often, however, in professional and consumer studios.

Tape Formats
There is a long list of video formats. Currently, there are two categories: composite and component. Within the composite category there are full-bandwidth composite recording systems and systems known as color-under. Color-under systems record the color signal at a frequency far beneath 3.58 MHz. One final distinction to be made is whether the format is digital or analog.

Here is a list of analog video formats and their relative qualities:
1" video tape: full-bandwidth composite video
BetacamSP: component Y, R-Y, B-Y video
Hi-8, SVHS: composite video
3/4" U-matic: composite color-under
VHS, Betamax: composite color-under

Digital formats:
D-1: component Y, R-Y, B-Y (sampled at 4:2:2 ratios)
D-2: composite video
D-3: ~~component Y, R-Y, B-Y~~ composite 1/2" size; Panasonic
Digital BetacamSP: component Y, R-Y, B-Y (sampled at 4:2:2, compressed 2:1)
D-5: component Y, R-Y, B-Y
DVCpro: compressed digital

There are no manufactured R, G, B component video tape formats.

There are a number of computer file formats which represent video and compression information:
JPEG
MPEG
TIFF
Quicktime
Avid
Quantel

All of these are digital files which can be stored on hard drives, backup tapes, CD-ROMs, or video tapes.

An Informal, Biased and Incomplete Survey of Tapestocks

Written by Paul Lundahl of Practical Productions

Pixelvision

Size: Identical to and interchangeable with a normal Phillips audiocassette.
Company: Fisher-Price Toy Company
Brief Description: Exclusively an acquisition format, black and white only, with a maximum recording time of 7 and a half minutes per side of a 90 minute audiotape. Only Camera for the format is a child's toy which is no longer manufactured in this country. Unlimited world wide format availability. Cassettes easy to smuggle in and out of politically repressive countries. Has a fiercely loyal following among fringe video artists due to its strange blocky low resolution image and economical format. Offers the ability to record video over all the boring audiotapes in your music collection.

VHS

Size: This is the kind of tape you rent at Blockbuster which is the "coin of the realm" standard of the video world.
Company: JVC
Brief Description: Through brilliant marketing and excellent luck VHS won out as the current U.S. and European playback standard over Sony's Beta format which is quite a bit superior. Used as an acquisition format mostly by tourists who want to pop out the unedited tape and play it right away in their home VCR's. Many incredible programs have been shot and edited on this format, however, precisely because of the economy of scale resulting from its widespread use. Also very popular as an off-line "roughcut" format because of it's inherant low cost and inexpensive, flexible editing equipment. Almost everything produced almost anywhere ends up on VHS at some point.

Super VHS

Size: Same exact size shell and tape as regular VHS only with differant recognition holes on the bottom and a higher quality tape formulation.
Company: JVC
Brief Description: Released shortly after Sony released H-8, its souped up Video-8 format, JVC responded with Super VHS. Engineers at JVC boosted the frequencies at which the luminence and chrominance (light and dark and color) signals are recorded. Super VHS is an excellent acquisition format with an image superior in the first generation to 3/4". Super VHS and Hi-8 are battling it out right now for the hearts and minds of independent producers. A wide range of editing options and lower dropout count have made it a stronger choice for TV stations and "industrial producers". There is also a comfortable familiarity with it having the same shape as the ubiquitous VHS, they fit together nicely on the same shelves and you can use Super VHS equipment to record and playback regular VHS. Since the equipment is large and clunky Super VHS is thought of as kind of nerdy.

Video-8

Size: A little bit smaller and a bit thicker than a regular Phillips Audiocassette.
Company: Sony
Brief Description: The first 8mm video format marketed by Sony as a consumer camcorder format. Sony chose the size as a shameless reference to Super-8 film which it attempts to be an evolution of. It has become the favorite new camcorder format recently surpassing VHS in sales and has many interesting capabilities which Sony pioneered. Video-8 has a noticeably better picture than VHS which has a cassette over 6 times larger. Video-8 also has 2 channels of digital audio recording, 2 channels of Hi-Fi audio recording and a seperate 24 bit futuristic timecode track which products haven't even been designed for yet. In the past few years Sony has been buying up movie companies like Columbia so that they can release feature films in Video-8. In places like Tokyo people watch feature films on little Video-8 Watchmans, a whole collection of 2 hour feature films can fit in a lunchpail. Because of its small size Video-8 is an excellent archival format. Political activists are very fond of the 8mm format because of the discrete size of the cameras, when Gorbachev was abducted he communicated to the world by a tape produced on a Video-8 camcorder and smuggled past his captors.

Hi-8

Size: Identical to Video-8 with differant equipment recognition holes and a higher quality tape formulation as in the case of Super VHS.
Company: Sony

Brief Description: Released a year after Video-8, Hi-8 was another extraordinary engineering achievement. With the same audio and Timecode capabilities as Video-8 the Sony engineers increased the frequency of the luminence and chrominence signals to create a format with superior image quality to 3/4", which for many years was the broadcast standard. With its small size and broadcast quality specs it has been used extensively by journalists and was often intercut with footage shot on $40,000 cameras during the gulf war. Very popular with filmmakers who held out from producing their work on video for aesthetic reasons but were converted to Hi-8 by: the size of the tapes, the excellent image quality, the economy, and a odd film-like quality to the image resulting mostly from the type of cameras that have been manufactured. (note: the 8mm formats are the favorites of the author of this survey who is very biased). Currently subject to a higher dropout count than other formats due to extremely fine tolerances which are needed in the manufacturing process.

Beta

Size: About 10 to 15% smaller than a standard VHS tape.
Company: Sony
Brief Description: Was the first consumer format and still popular in a retro fringe community. Beta (or Betamax) remains a popular format in Mexico and Central America. Pretty much obsolete in the United States.

ED Beta

Size: Same size tape as the original Beta.
Company: Sony
Brief Description: A very interesting format which is similar in image quality to Hi-8 and Super VHS but maybe a little better. Marketed mostly to football Coaches at High Schools and Universities as a replacement for the 16mm Black and White film they had been using to document practices and critique performance. Sony did not support it with editing equipment or tapestock selection and it has become an obsolete format even while many people are still using it.ED Beta products are no longer in production at least for U.S. distribution. You can use 30 minute Beta SP tapes in ED Beta equipment but due to the slower recording speed they hold 3 hours of material!

U-Matic or 3/4"

Size: Large cassettes about 1/2 the size of the San Francisco white pages.
Company: Sony
Brief Description: U-matic tape dominated the professional video production industry for many years, professional facilities still use 3/4" for editing or submastering. U-matic can be thought of as the VHS of the professional video world. U-matic has the widest range of editing equipment built for it and since the tape is so wide it is very stable in recording and playback. Since the image quality is inferior to the more economical Hi-8 and Super VHS, 3/4" is becoming known as a dinosaur format.

U-Matic SP or 3/4" SP

Size: Identical to regular 3/4"
Company: Sony
Brief Description: In a successful attempt to keep 3/4" alive, Sony increased the frequency of the luminence and chrominence signals of 3/4" SP in a similar method to Hi-8 and Super VHS. Since facilities could still play all of their regular 3/4" material on the new 3/4"SP decks it has become a very popular editing format, though used less often for acquisition.

1" Type C

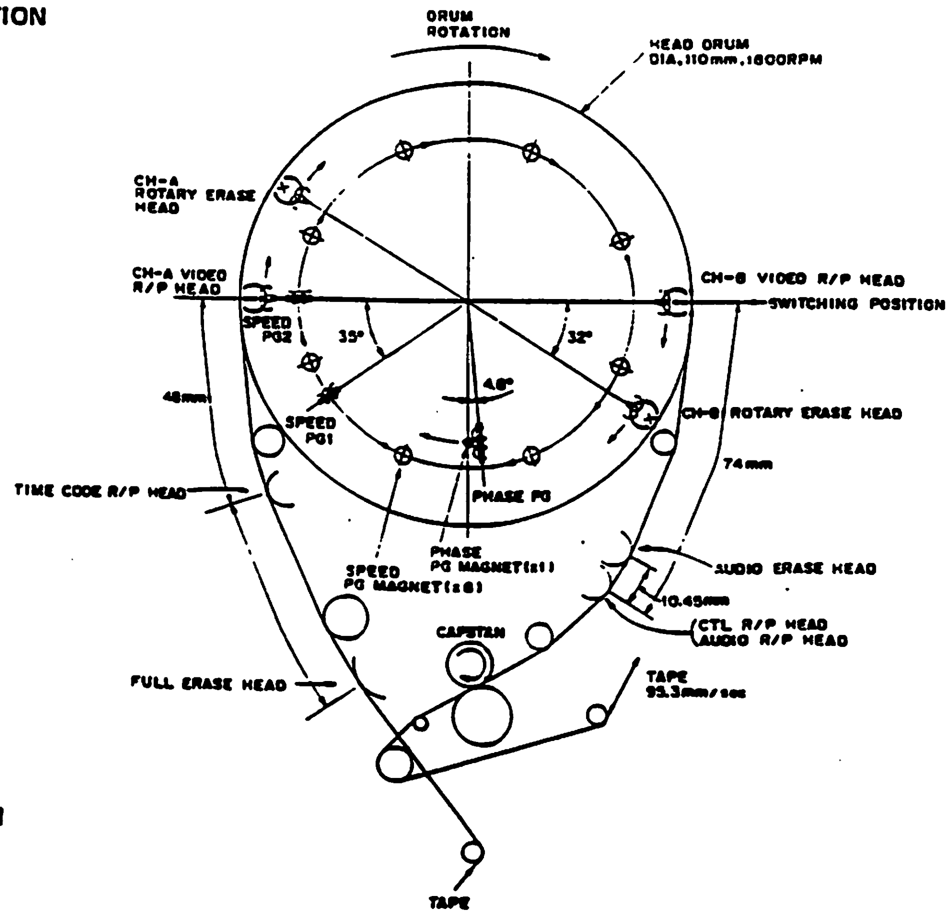
Size: Open reel Spools of varying sizes depending on the length, 1" thick.
Company: Ampex
Brief Description: Used primarily as a format to master to for broadcast, 1" was the "ultimate" format since the early 60's when it was introduced, till the early 80's when component Betacam grew to dominate. Because of the awkwardness of its use (having to be laboriously threaded by hand), its high cost, and competition from cheaper higher resolution formats, 1" is disappearing from the scene.

U-MATIC 3/4" 149

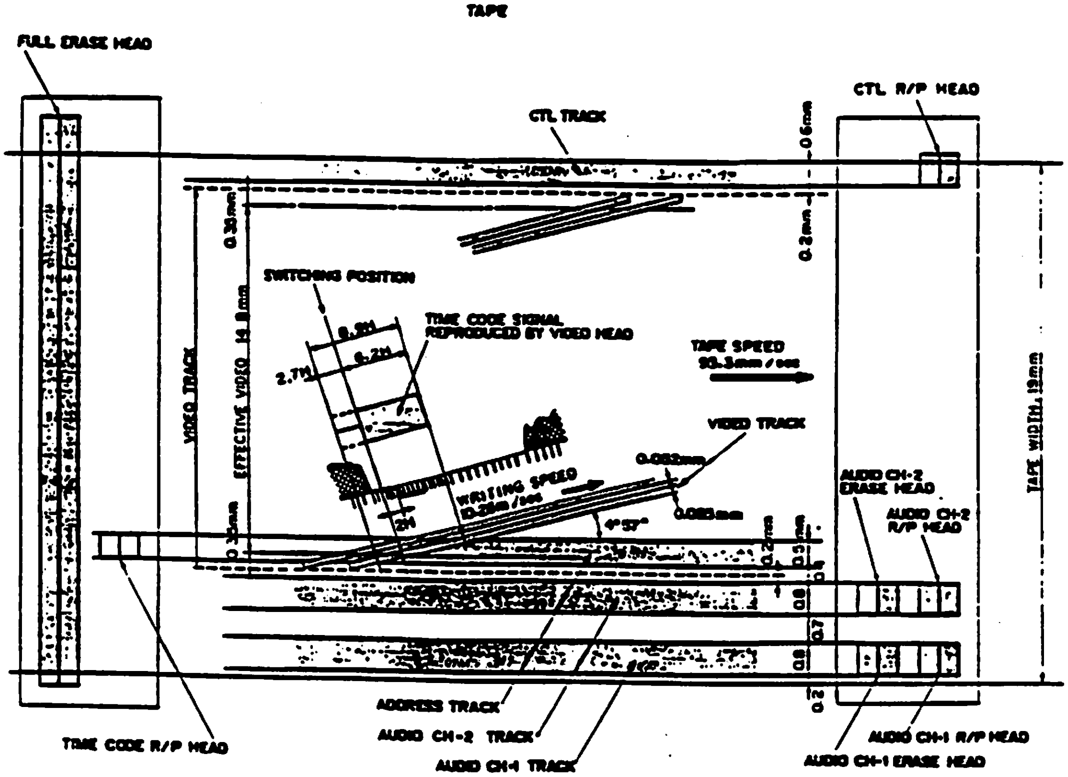
BLOCK DIAGRAM

TAPE FORMAT

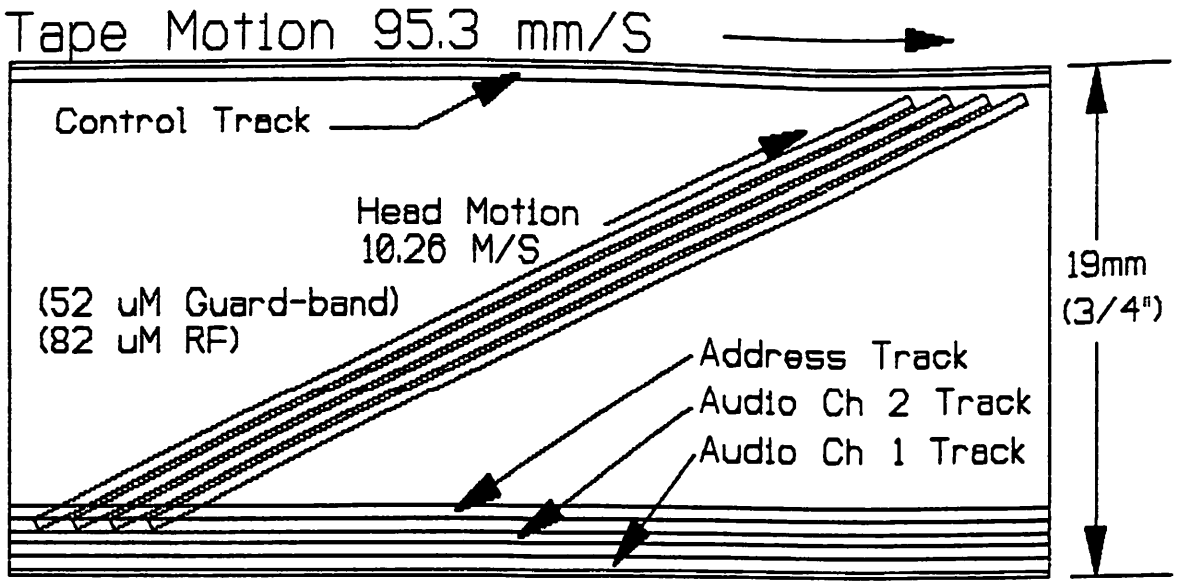
HEADS LOCATION



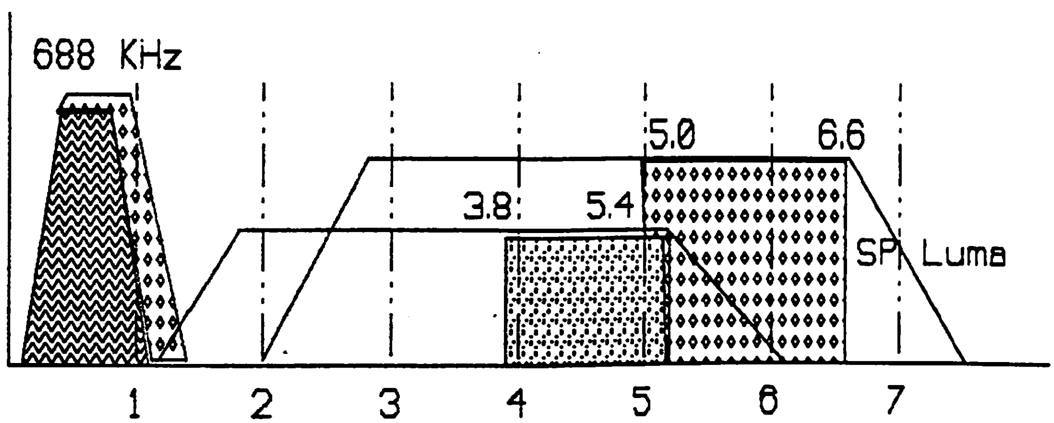
TAPE PATTERN



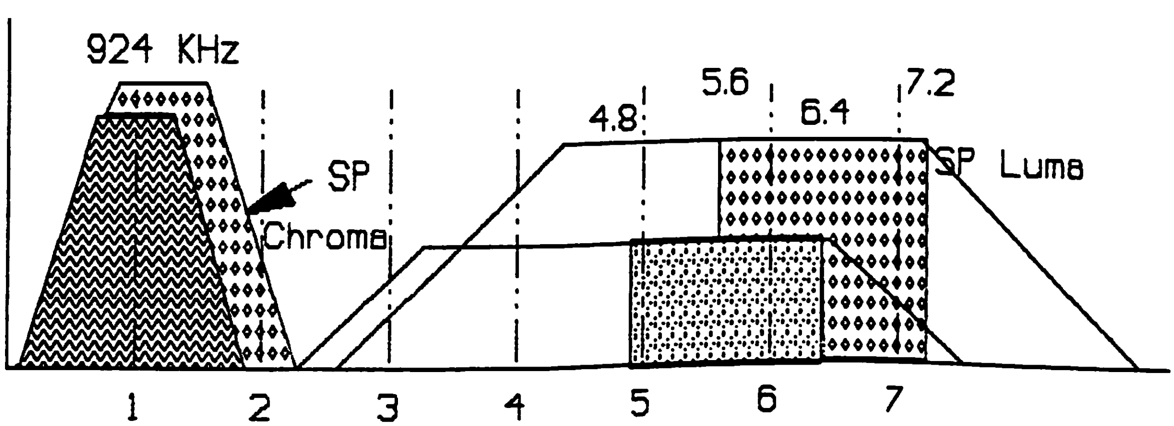
U-MATIC (3/4") FORMAT



NTSC Industrial & Broadcast

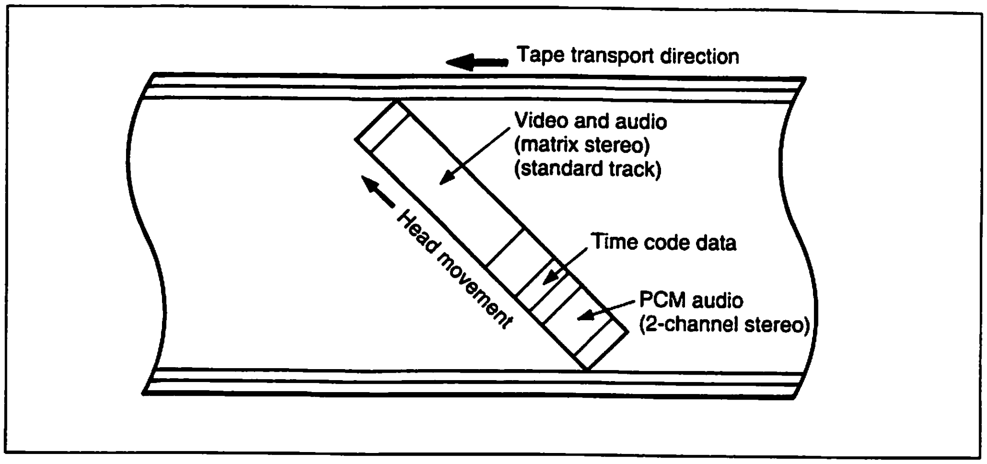


PAL Broadcast Only - Industrial is 688



8-mm format video cassette tape

The video and audio information are recorded on an 8-mm format video cassette tape according to the following allocation. This unit allows recording of the time code as well as other information.



8-mm format video tape

Recording and playback with a Hi8 cassette tape

A Hi8 metal tape with large magnetic energy allows high-density recording, and makes it possible to record and play back a high-quality picture.

Cassette tape and automatic switching of recording/playback mode

When using a Hi8 cassette tape for recoding, this unit senses the detection holes on the cassette shell, and automatically performs recording in the SP (standard play) mode of Hi8 video system.
When using a conventional 8-mm tape, the recording is performed in the conventional 8-mm video system.
In playback, the unit can detect the system mode used in recording by verifying the recorded signal, and plays back the tape in appropriate mode.

JVC unveils Super VHS (S-VHS) technical specs

JVC has disclosed final technical specifications for the Super VHS (S-VHS) technology the company announced in early January. S-VHS technologically allows the realization of horizontal resolution of more than 400 lines, making for a significant picture-quality improvement in the VHS format.

Since the announcement of S-VHS, the technology has been the focus of growing attention. In collaboration with Hitachi Ltd., Matsushita Electric Industrial Co., Ltd., Mitsubishi Electric Corp., and Sharp Corp., JVC has set the format's technical points as follows:

1. S-VHS Cassette Tape:

- *Cassette shell:* VHS cassette with an ID (identification) hole which is already specified as a spare hole in the original VHS standard
- *Tape:* Higher-performance tape. The new reference tape is defined

2. Recording Modes:

- *SP mode:* tape speed—3.3 cm/sec.; recording time—120 minutes, with a tape length equivalent to that of a conventional T-120 VHS cassette
- *EP mode:* tape speed—1.1 cm/sec.; recording time—360 minutes, with a tape length equivalent to that of a conventional T-120 VHS cassette

3. Video Signal Recording Method:

- *Video input/output signal:* NTSC signal or separated YC signals conforming to NTSC signals
- *Luminance signal recording:* Frequency modulated (FM) recording

FM signal frequency:
White peak level frequency . . . . 7.0 MHz
Sync tip level frequency . . . . 5.4 MHz
Frequency deviation . . . . 1.6 MHz
Clip level:
White clip 210%
Dark clip 70%
Emphasis:
Main emphasis . . . . VHS standard emphasis
Sub emphasis . . . . non-linear emphasis:

- *Chrominance signal recording:* Down-converted direct recording

4. Audio Signal Recording Methods:

- *Linear track recording method:* conforming to VHS audio recording method
- *FM recording method:* conforming to VHS audio recording method

S-VHS backgrounder

Following is some explanatory information on Super VHS technology and its effects on picture improvement:

1. High Performance S-VHS Cassette with ID Hole:

The Super VHS system uses a high-performance S-VHS cassette based on new specifications. Improved tape quality has contributed largely to the achievement of higher picture quality.

An S-VHS cassette shell has an ID (identification) hole to allow an S-VHS VCR, which is equipped with both Super VHS and conventional VHS recording modes, to automatically distinguish between S-VHS and conventional VHS cassettes, thus preventing inadvertent recording in the S-VHS mode when using a conventional VHS cassette.

2. 7MHz (white peak level) FM Signal Recording to Realize High Resolution:

Although Super VHS will use the same frequency-modulated recording method as used in the conventional VHS format for luminance signal recording, the FM frequency range has been changed from the conventional format's 3.4MHz–4.4MHz to 5.4MHz–7.0MHz. Frequency deviation has also been changed from conventional VHS's 1MHz to 1.6MHz. These changes contribute to realizing horizontal resolution of more than 400 lines.

3. Separated YC Input/Output Signals to Achieve Picture Quality Improvement:

To achieve overall high picture quality, for video input/output signals Y (luminance) and C (chrominance) signals are used in addition to NTSC signals currently used, in order to eliminate interference between luminance and chrominance signals.

4. Forward (continuous) Compatibility between VHS and Super VHS:

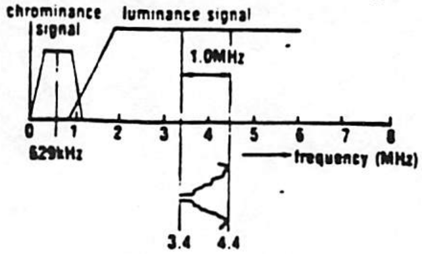
Super VHS has the same tape speeds as the conventional VHS standard: 3.3cm/sec. in the SP mode and 1.1cm/sec. in the EP mode. When an S-VHS cassette that has a tape length equivalent to that of a conventional T-120 VHS cassette is used, record/playback time is 120 minutes in the SP mode and 360 minutes in the EP mode.

In recording audio signals, the AC-bias recording method is used for linear track recording, and frequency-modulated depth multiplex recording (D-MPX) is used for recording Hi-Fi audio signals.

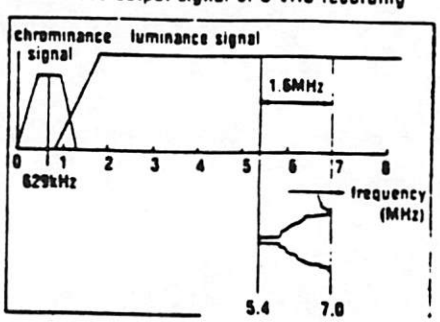
S-VHS VCRs will be equipped with recording modes for both conventional VHS and S-VHS systems, to enable users to select either mode. Also an S-VHS cassette can be used with any conventional VHS VCR, ensuring format continuity between conventional VHS and Super VHS. ■

Recording Signal Spectrum

Video head output signal of VHS recording



Video head output signal of S-VHS recording



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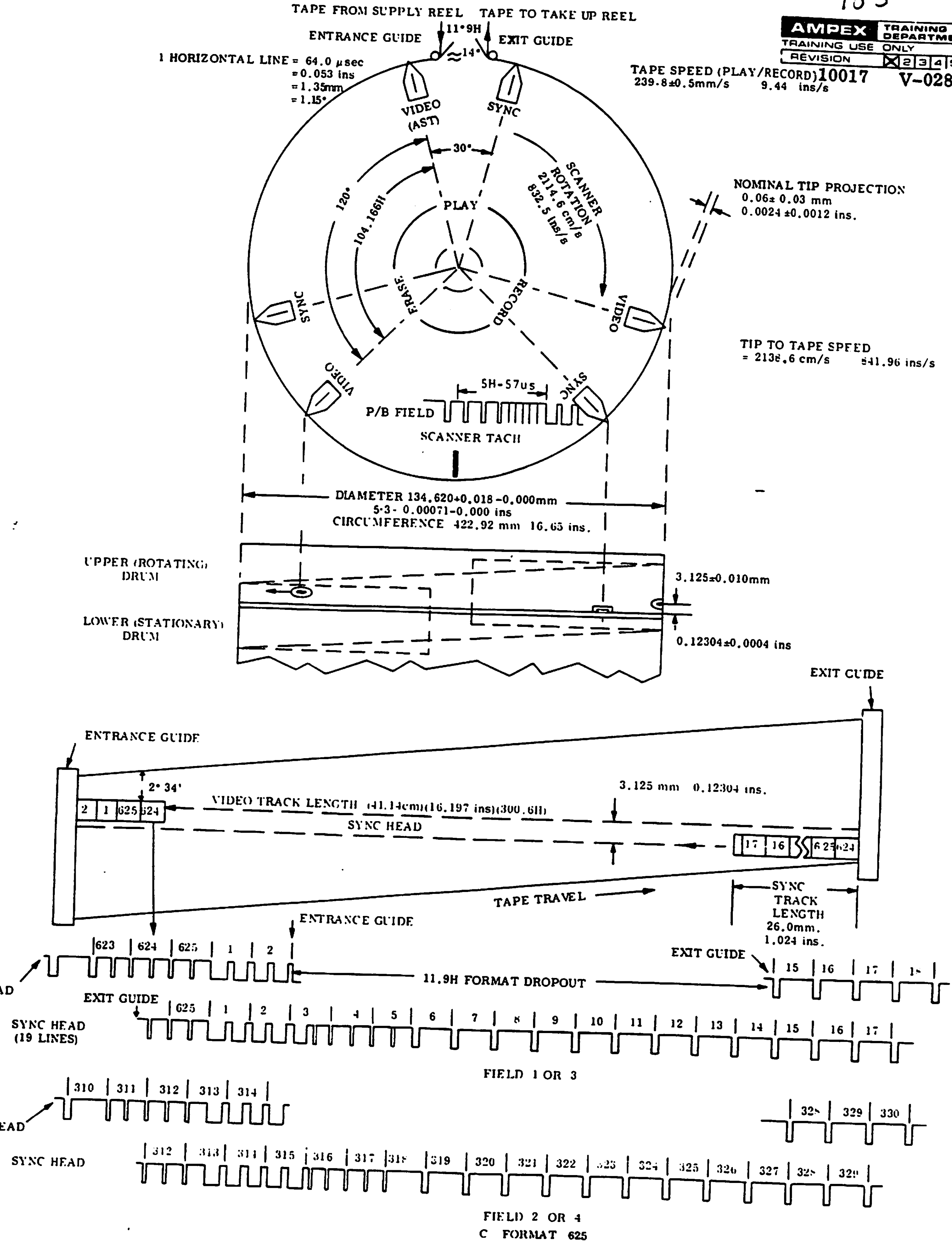
TAPE FROM SUPPLY REEL
TAPE TO TAKE UP REEL
11°9H
ENTRANCE GUIDE
EXIT GUIDE
14°
1 HORIZONTAL LINE = 64.0 µsec
= 0.053 ins
= 1.35mm
= 1.15°
TAPE SPEED (PLAY/RECORD)
239.8±0.5mm/s 9.44 ins/s
VIDEO (AST)
SYNC
30°
120°
104.166H
PLAY
SCANNER ROTATION 2114.6 cm/s 832.5 ins/s
NOMINAL TIP PROJECTION
0.06± 0.03 mm
0.0024 ±0.0012 ins.
ERASE
RECORD
SYNC
VIDEO
VIDEO
SYNC
TIP TO TAPE SPEED
= 2136.6 cm/s 841.96 ins/s
5H-57us
P/B FIELD
SCANNER TACH
DIAMETER 134.620+0.018-0.000mm
5·3- 0.00071-0.000 ins
CIRCUMFERENCE 422.92 mm 16.65 ins.
UPPER (ROTATING) DRUM
LOWER (STATIONARY) DRUM
3.125±0.010mm
0.12304±0.0004 ins
EXIT GUIDE
ENTRANCE GUIDE
2° 34'
VIDEO TRACK LENGTH (41.14cm)(16.197 ins)(300.6H)
3.125 mm 0.12304 ins.
SYNC HEAD
TAPE TRAVEL
SYNC TRACK LENGTH
26.0mm.
1.024 ins.
ENTRANCE GUIDE
623 624 625 1 2
HEAD
11.9H FORMAT DROPOUT
EXIT GUIDE
15 16 17 18
EXIT GUIDE
625 1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16 17
SYNC HEAD
(19 LINES)
FIELD 1 OR 3
310 311 312 313 314
HEAD
328 329 330
SYNC HEAD
312 313 314 315 316 317 318 319 320 321 322 323 324 325 326 327 328 329
FIELD 2 OR 4
C FORMAT 625

154

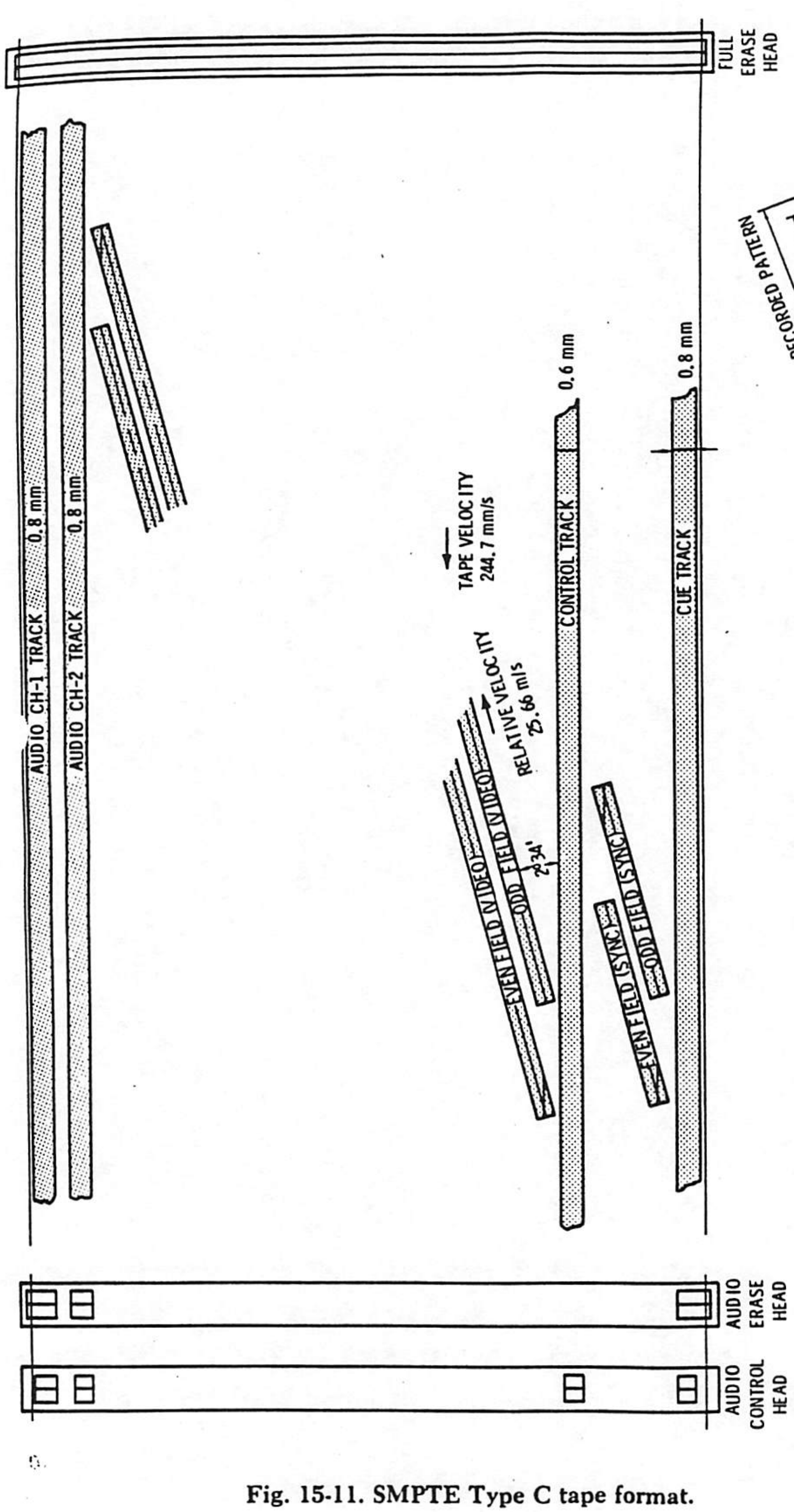

Fig. 15-11. SMPTE Type C tape format.

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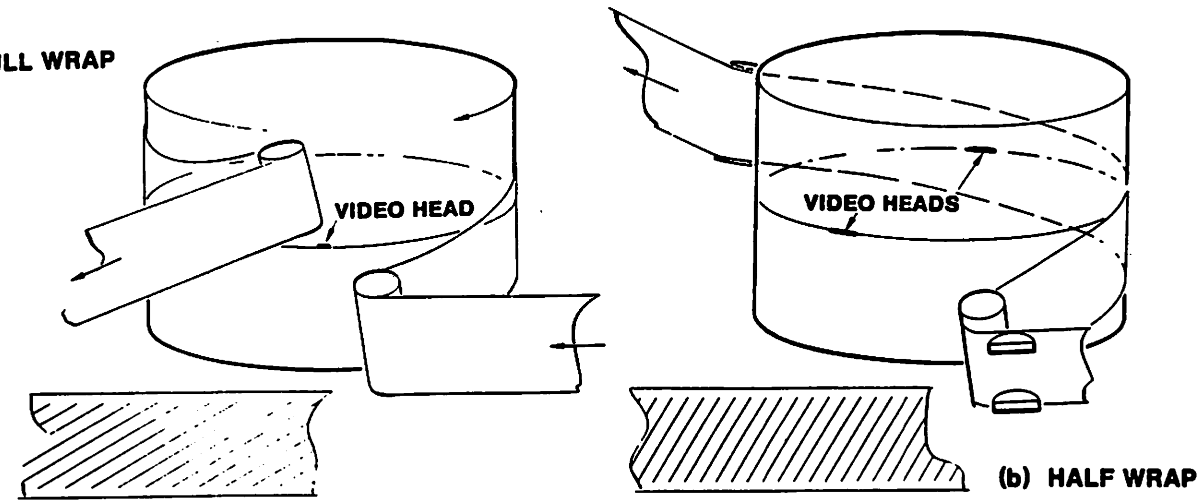


Fig. 62 Helical Configurations and Video Track Patterns

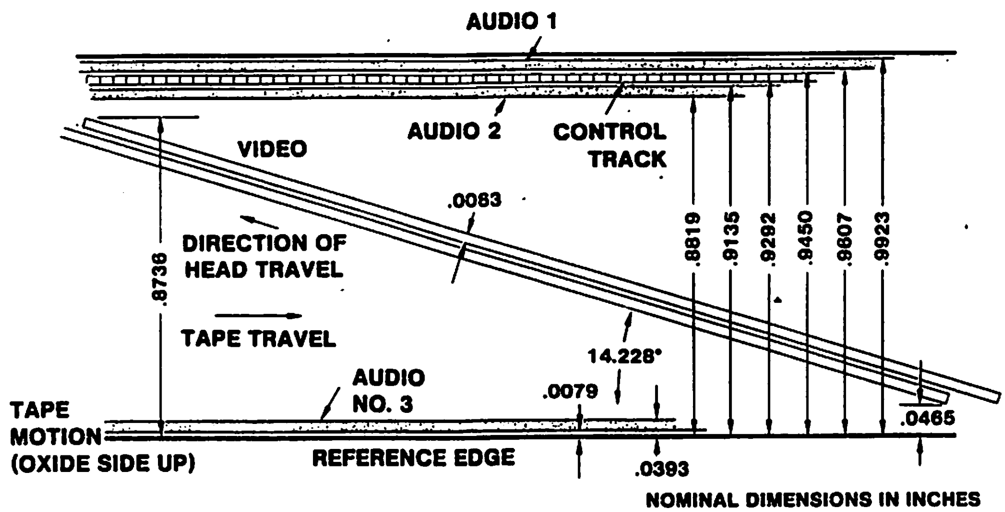


Fig. 63 Type B Format

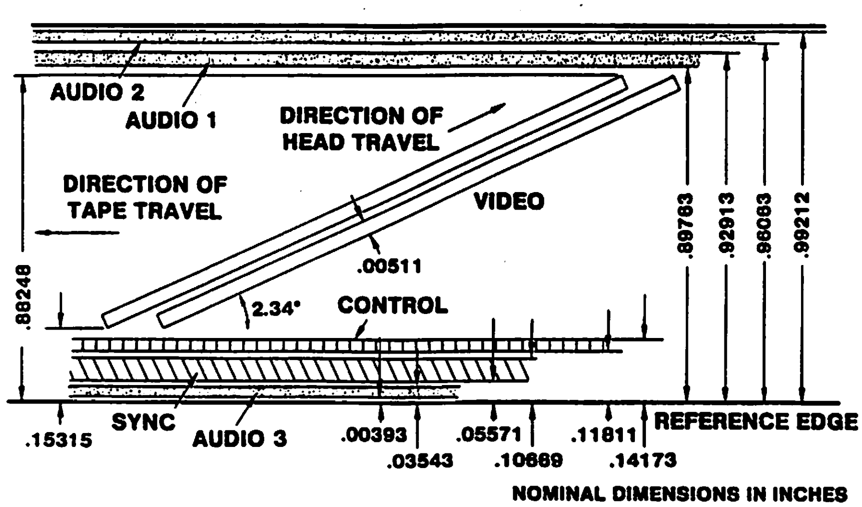
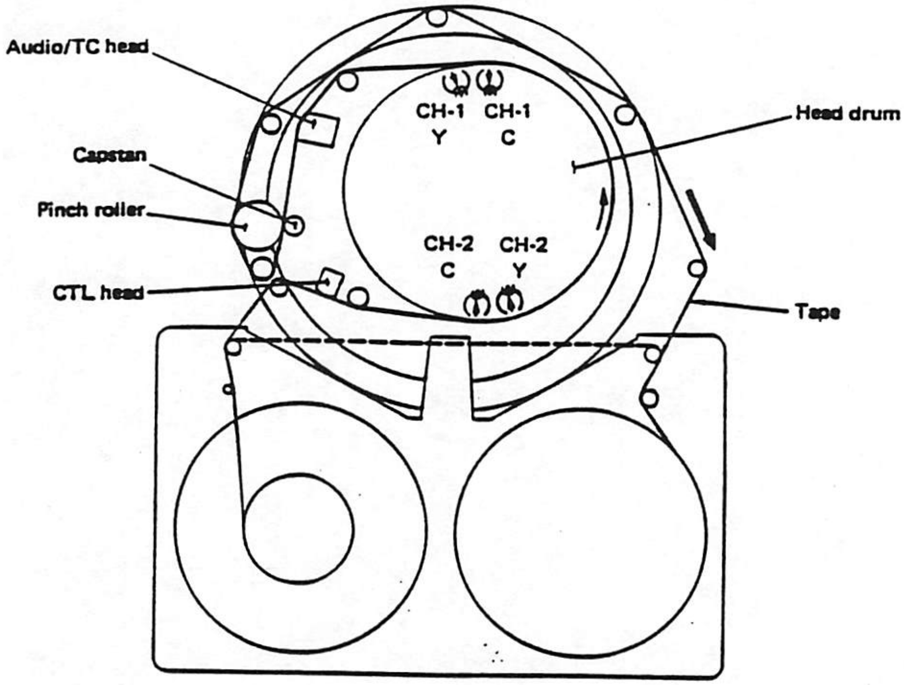


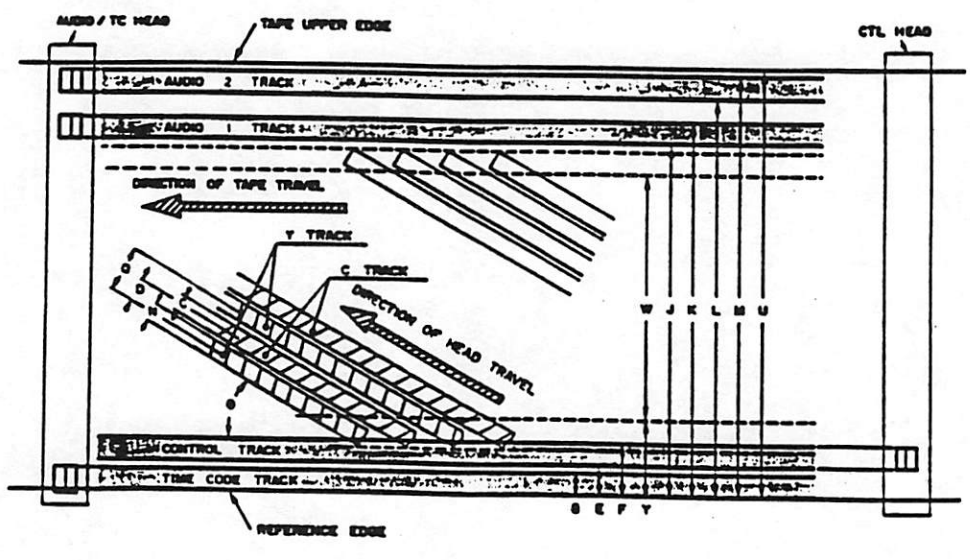
Fig. 64 Type C Format

BETACAM $\frac{1}{2}$" 156

TAPE TRANSPORT



TAPE PATTERN



B :Time Code Track Upper Edge	0.4	L :Audio 2 Track Lower Edge	11.85
C :C Track Width	0.073	M :Audio 2 Track Upper Edge	12.45
D :Y-C Track Pitch	0.0805	N :Y Track Width	0.073
E :Control Track Lower Edge	0.7	Q :Video Track Pitch	0.161
F :Control Track upper Edge	1.1	U :Tape Width	12.7
J :Audio 1 Track Lower Edge	10.85	W :Video Area Effective Width	9.384
K :Audio 1 Track Upper Edge	11.45	Y :Lower Limit of W	1.248
		θ :Track Angle	4.679°

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6. FORMAT STANDARD

Digital Betacam

1. Format Specifications

The Digital BETACAM format conforms to the video signal encoding parameters of the CCIR-601 Recommendation adopted for the D-1 format, but extended for 10-bit quantization. Therefore, a 4:2:2 digital component video signal can be directly interfaced to Digital BETACAM VTRs via their digital input and output ports.
However, signal recorded on tape is completely different from that of the D-1 format because of the highly efficient data handling system (Bit Rate Reduction Scheme for video signal) used in Digital BETACAM.

A/D conversion of the optional analog composite video input and D/A conversion of the analog component and composite video outputs are executed at 10 bits per sample. A/D conversion of the analog component video input is carried out at 8 bits per sample.

Digital BETACAM video encoding parameters

Sampling frequency	Y: 13.5MHz R-Y: 6.75MHz B-Y: 6.75MHz
Quantization	10-bit (8-bit A/D quantization for analog component input)
Number of samples/ horizontal line	Y: 858 samples R-Y: 429 samples B-Y 429 samples
Number of samples/ digital active line	Y: 720 samples R-Y: 360 samples B-Y: 360 samples -------------------- Total: 1440 samples
Recorded lines/field	256 lines

Table 2

The Digital BETACAM format includes four 20-bit AES/EBU digital audio tracks which are independently editable. While 20-bit digital audio signals can be interfaced through the digital I/O, A/D and D/A conversion of analog input and output audio signals is carried out at 18 bits per sample.

Digital BETACAM audio encoding parameters

Sampling frequency	48kHz (synchronized with video)
Quantization	20-bit (18-bit A/D and D/A conversion)

Table 3

2. Tape and Cassette

A half-inch metal particle tape has been specially developed for the Digital BETACAM format to suit the much narrower track width and shorter wavelength of high density digital recording. Its superfine magnetic particles and hyper molecular binder give a high RF output and a hyper lubricant protects the tape from the increased stress of a larger number of heads and a writing speed three times higher than the Betacam/Betacam SP format.
While the cassette shell was designed to have the same dimensions as current Betacam/Betacam SP cassettes for playback compatibility, several ID holes are incorporated which confirm that the recording is digital, the tape is metal and its thickness is 14μm.
To prevent dust particles from accumulating on the tape, which could cause drop-outs during recording and playback, Digital BETACAM cassettes have an anti-static lid and dust-proof case. This dust protection design, combined with the dust protection system of the VTR tape transport, helps to ensure accurate recording of digital data.

Digital BETACAM tape specifications

Tape width	12.65mm (1/2-inch)	
Thickness	metal coating	3.0μm
	base	10.0μm
	backcoating	1.0μm
	Total	14.0μm
Material	Metal particle	

Table 4

Digital BETACAM cassette dimensions

Small cassette (WxHxD)	156 x 96 x 25mm
Large cassette (WxHxD)	245 x 145 x 25mm

Table 5

Digital BETACAM cassette running times

Small cassette	40 min
Large cassette	124 min

Table 6

3. Scanner

To obtain good head to tape contact,
Digital BETACAM VTRs have a mid-rotary scanner which is positioned between fixed upper and lower drums. The number of heads on the scanner depends on whether the model is a recorder or player and if it has a Betacam/Betacam SP playback capability. [Fig.14] shows the scanner layout of the DVW-A500, which has the most heads of the four models.

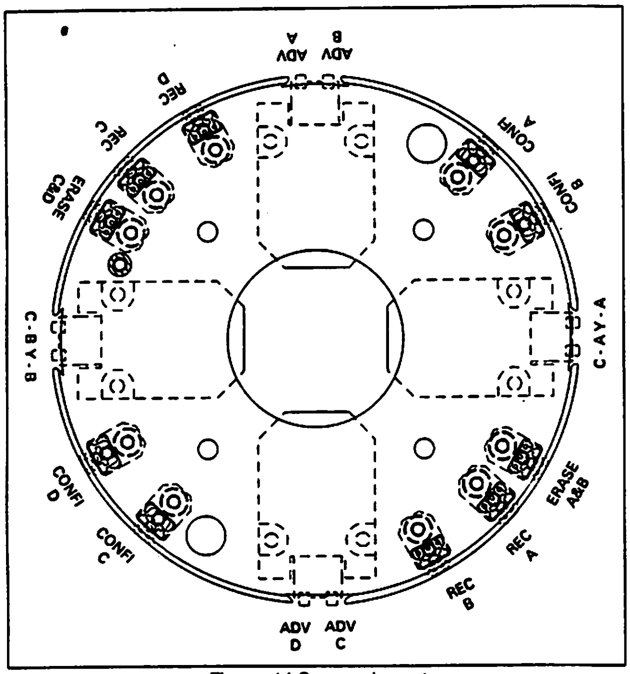


Figure. 14 Scanner Layout

A total of 18 heads are attached to this scanner, including 4 recording heads, 2 flying erase heads, 4 advanced DT heads, 4 confidence heads and another 4 DT heads for Betacam/Betacam SP playback. These groups of four heads (except for the erase heads) are respectively coupled into pairs and the two pairs of each group are located 180° apart from each other. Since the effective wrap angle of the tape is a little less than 180°, only one of each of the two pairs has contact with the tape at one time. Hence, two tracks are recorded or played back together at every half rotation of the scanner, and four tracks are recorded or played back with every complete rotation. Each of the two flying erase heads erases two tracks at any one time.

4. Format Footprint

[Fig.15] shows the foot print of the Digital BETACAM format.
As shown in the figure, six helical tracks contain the video and audio data corresponding to one field. Each helical track contains two video sectors and four audio sectors. Each of the four audio sectors corresponds to one of four digital audio channels (CH.1 to CH.4). [Fig.15] shows the allocation of digital audio data to each sector within a field. Additionally, on the A-channel track, two kinds of tracking pilot signal are recorded in the two edit gaps between the video and audio sectors respectively and one of them is also recorded in the upper gap of the C-channel track.

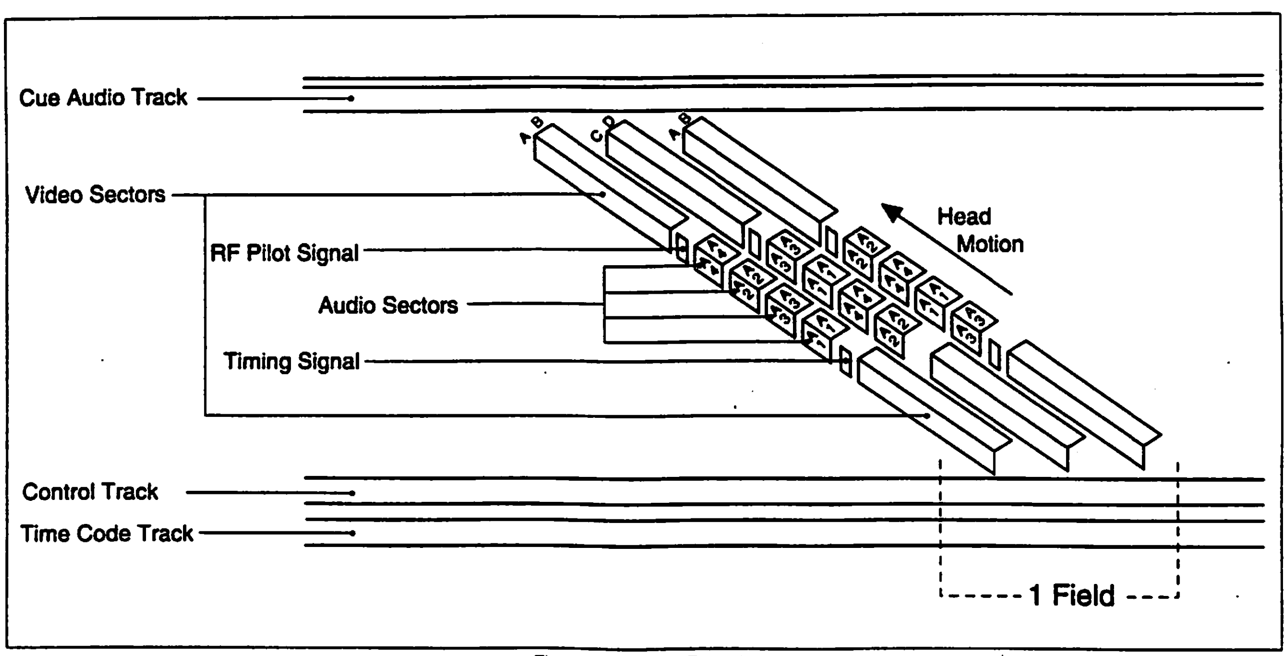


Figure. 15 Format Footprint

5. Pilot Tone Tracking System

The structure of a digital signal is such that it can have a relatively poor signal-to-noise ratio but still allow complete data recovery. In turn, this allows a digital VTR to operate with a much narrower track width than its analog equivalent. In the Digital BETACAM format the track width is only 20μm and, while it is possible for a CTL-based servo system to provide the required tracking accuracy, nevertheless it could require frequent adjustment in day to day operations which involve frequent tape interchanging.

Digital BETACAM VTRs therefore have a pilot tone tracking system in which two pilot tracking signals are recorded in gaps between the video sectors and audio sectors within certain channels of the helical tracks, as shown in [Fig.8]. In the upper gap, a 4MHz RF pilot signal is recorded and a 400kHz timing signal is recorded in the lower gap.

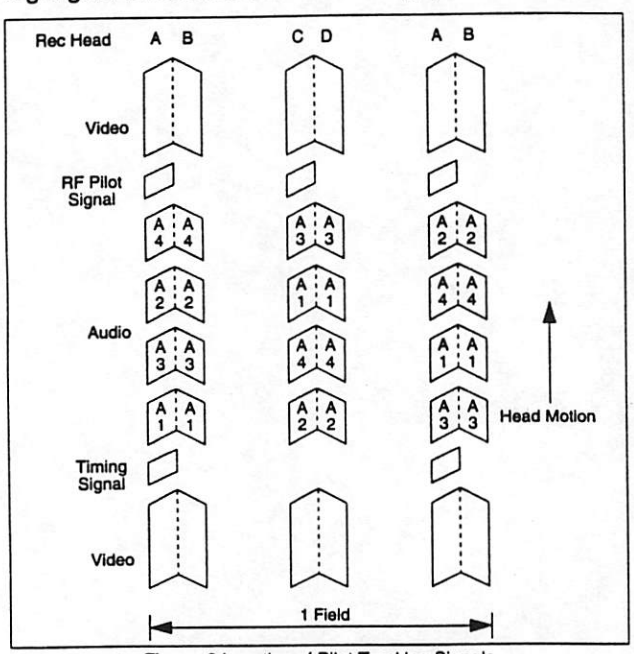


Figure. 8 Location of Pilot Tracking Signals

The 400kHz timing signal is used to correct longitudinal tracking errors and is detected with the A-channel record head. Its timing is compared to that of the pulse signal derived from the frequency generator within the drum, and the time difference between these signals correlates to the deviation from the correct tracking path. This is shown in [Fig.9]. A servo CPU calculates the deviation from the time difference, and controls the capstan drive so that tracking is optimized. This process is executed during the servo lock-up time whenever an edit or playback is started.

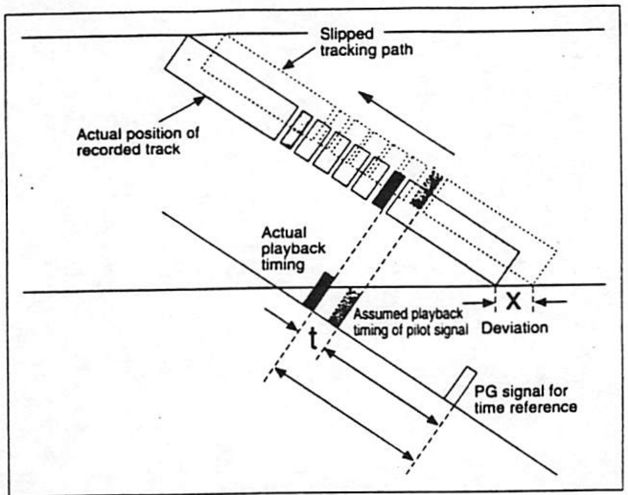


Figure. 9 Pilot Tone Tracking

When editing is executed, the newly recorded tracks must be in their proper phase relationship to those previously recorded so that these existing tracks are not damaged at the editing point and tracking errors do not occur in playback. Therefore, more precise tracking is needed for editing and the 4MHz RF pilot signal mentioned above is used to further improve tracking accuracy.

During preroll, the A and C-channel recording heads detect the RF pilot signal on tape and output the RF envelope to the capstan servo CPU. As shown in [Fig.10], the CPU controls the capstan drive in the direction that increases the RF level and detects the point where the RF level is maximized. This feedback process is continuously repeated to keep the RF level at a maximum.

As with the CTL signal, these tracking reference signals are not erased with the flying erase heads during insert editing. An important advantage of the pilot tone tracking system is that, when insert editing, it continues to operate both during and after preroll until the complete edit is finished.

Thus, the pilot tone tracking system eliminates the need for manual tracking adjustment of the recording heads and improves editing efficiency.

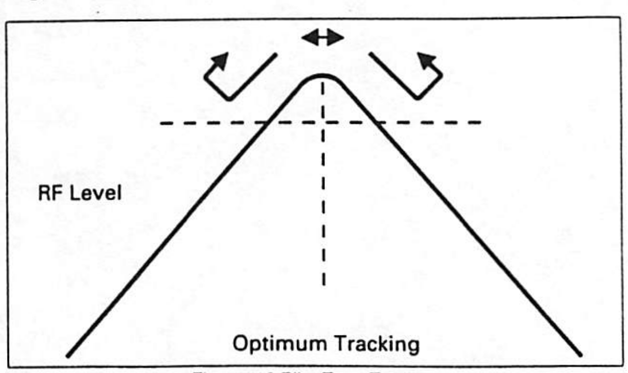


Figure. 10 Pilot Tone Tracking

Digital Betacam

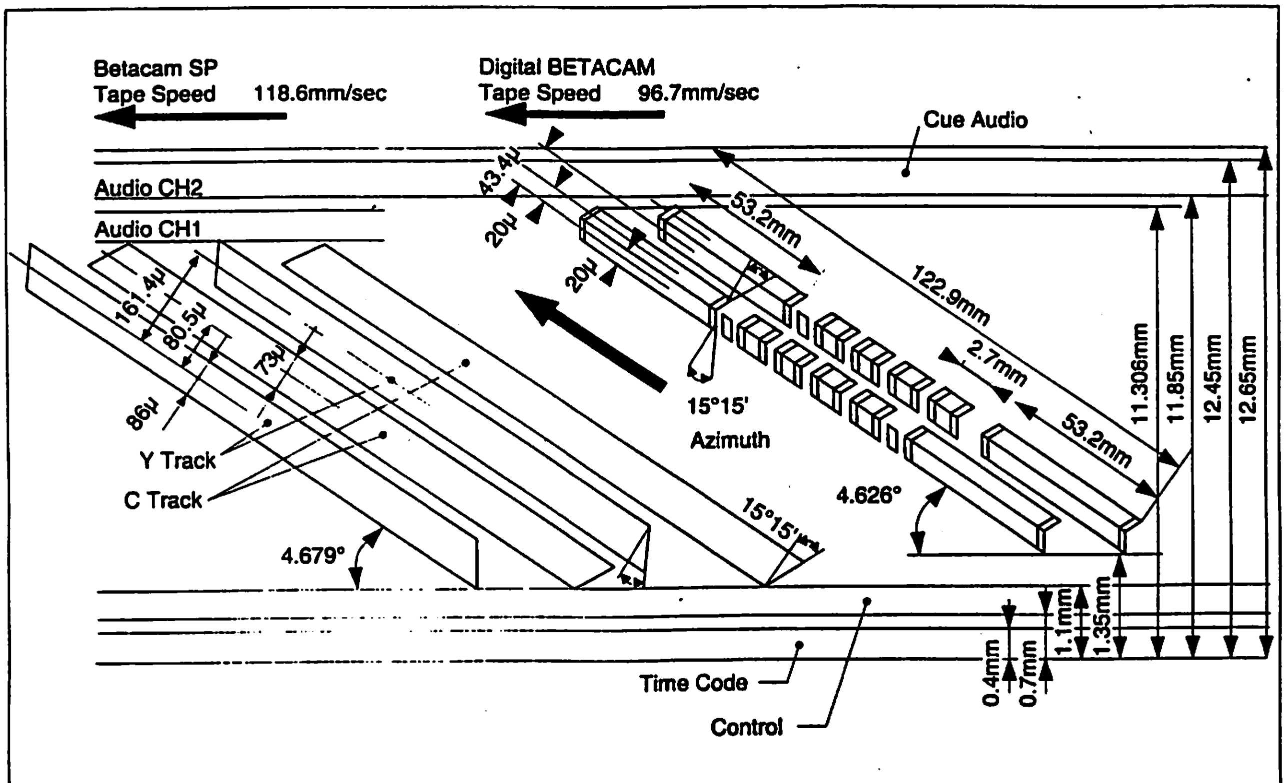


Figure. 16 Comparison of Digital BETACAM Format Footprint with Betacam SP

As shown in [Fig.16], three longitudinal tracks for cue audio, control and time code are recorded in the same position as those of the current Betacam/Betacam SP format so that analog Betacam/Betacam SP tapes can be played back with the same stationary heads for these signals. While the analog Betacam/Betacam SP format has two longitudinal audio tracks, Digital BETACAM has only a cue audio longitudinal track along the upper edge of the tape where channel-2 audio is recorded in the Betacam/Betacam SP format. Control and time code tracks are recorded in exactly the same position for both formats.

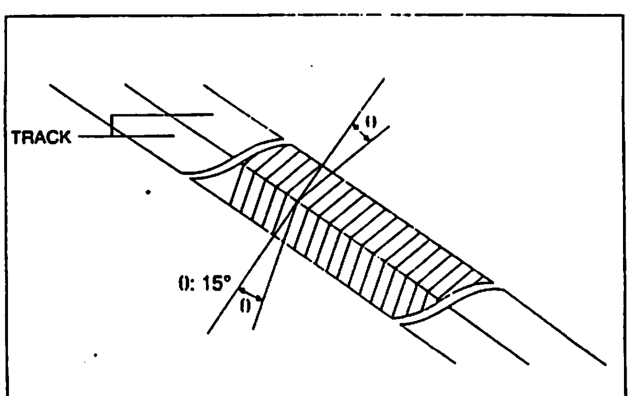


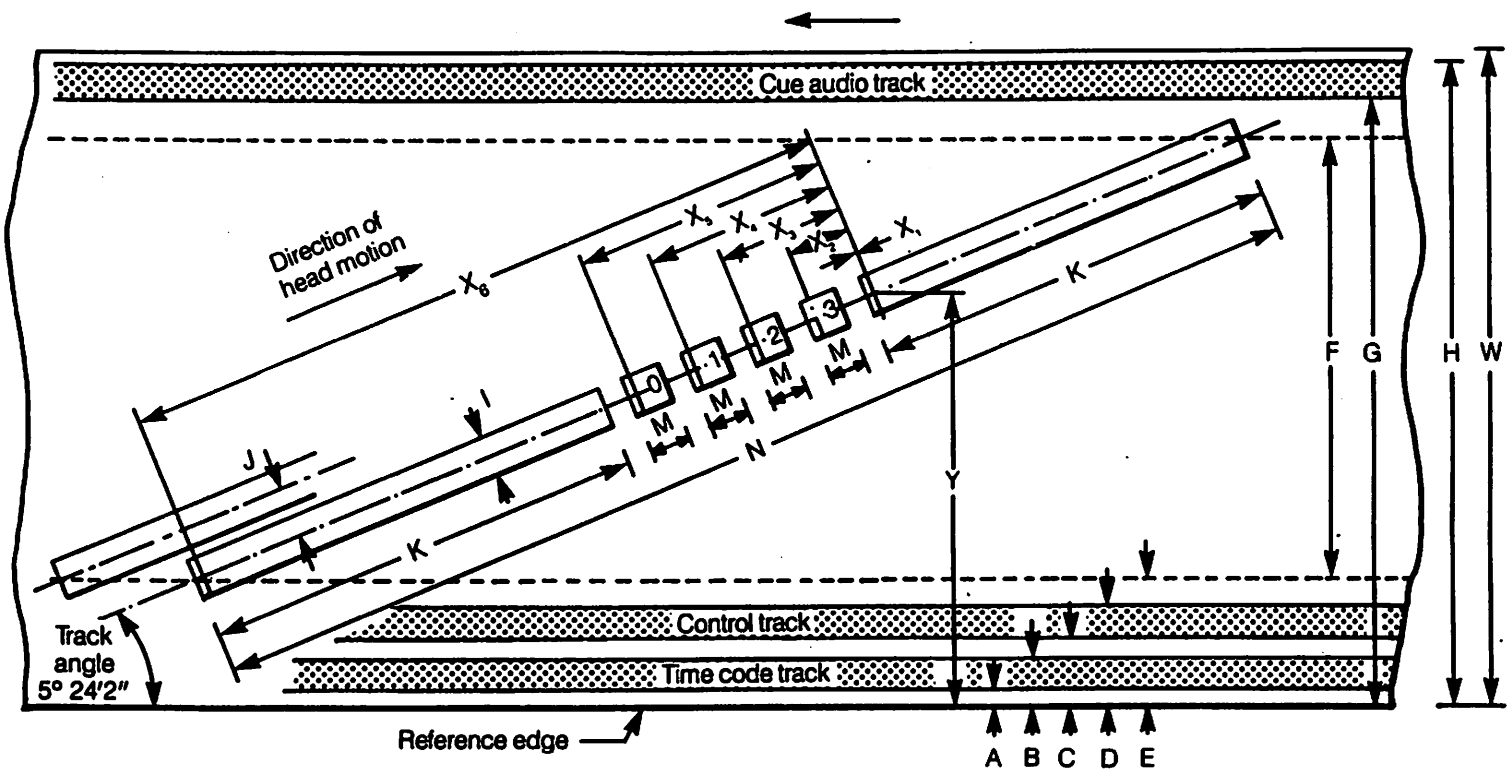
Figure. 17 Azimuth Recording

5. Azimuth Recording

In general, playback heads are designed to be wider than the actual track width, allowing a margin to accurately trace the narrow tracks. Theoretically, this wide playback head could cause crosstalk from adjacent tracks, resulting in a deterioration of the original signal, unless there was a guardband between adjacent tracks. However, it is obvious that this solution is not suitable for high density recording. In order to solve this problem the Digital BETACAM format uses an azimuth recording technique to achieve its high density recording with the precise reproduction of recorded signals. This azimuth recording method is well proven in Sony Betacam/Betacam SP and D-2 VTRs.

When helical tracks are recorded on the tape, the particles of the recording medium are magnetized with an azimuth angle of 15° to the perpendicular line of the tracking path, as shown in [Fig.17]. This angle is reversed on adjacent tracks so that crosstalk between adjacent tracks does not occur even though the playback head traces over both tracks. This is because each playback head has the same azimuth angle as the track to be reproduced, but has the opposite azimuth angle to the adjacent track.

D-1 FORMAT



A	Time-code track lower edge	0.2	±0.1
B	Time-code track upper edge	0.7	±0.1
C	Control track lower edge	1.0	±0.1
D	Control track upper edge	1.5	±0.05
E	Lower edge of programme area	1.8	(derived)
F*	Programme area width	16.00	(derived)
G	Cue audio track lower edge	18.1	±0.15
H	Cue audio track upper edge	18.8	±0.2
I	Programme track width	0.040	±0/-0.005
J	Programme track pitch	0.045	basic
K	Video sector length	77.78	(derived)
M	Audio sector length	2.56	(derived)
N*	Programme track total length	170.00	(derived)
W	Tape width	19.010	0.015

X_1	Location of start of upper video sector	0	±0.1
X_2	Location of start of audio sector 3	3.4	±0.1
X_3	Location of start of audio sector 2	6.8	±0.1
X_4	Location of start of audio sector 1	10.2	±0.1
X_5	Location of start of audio sector 0	13.6	±0.1
X_6	Location of start of lower video sector	92.2	±0.1
Y	Programme track reference location	10.490	basic

All dimensions in mm

* For 525/60; F: 16.00/1.001, N: 170.00/1.001

Fig. 5.1 Track pattern and dimensions

D-2 19mm Format

The overall location of tracks is shown in Figure 1-6. The only analog track is the cue track and it cannot be used as a off line editing track for audio since its output isn't in time with playback video due to a one field delay thru the processor in playback. The video will be output timed with house timing but the cue audio will be early. Figure 1-7 shows servo reference pulses are recorded to pulse doublet timing details at the SECTOR rate which is 180 HZ. Servo still needs Frame and Color Frame reference pulses which which are recorded at their usual 30 and 15 HZ rates respectively. Longitudinal Time Code is recorded below Ctl with conventional NRZ code. The VITC time code is incorporated into the digital Video data on the Sector tracks with the video data.

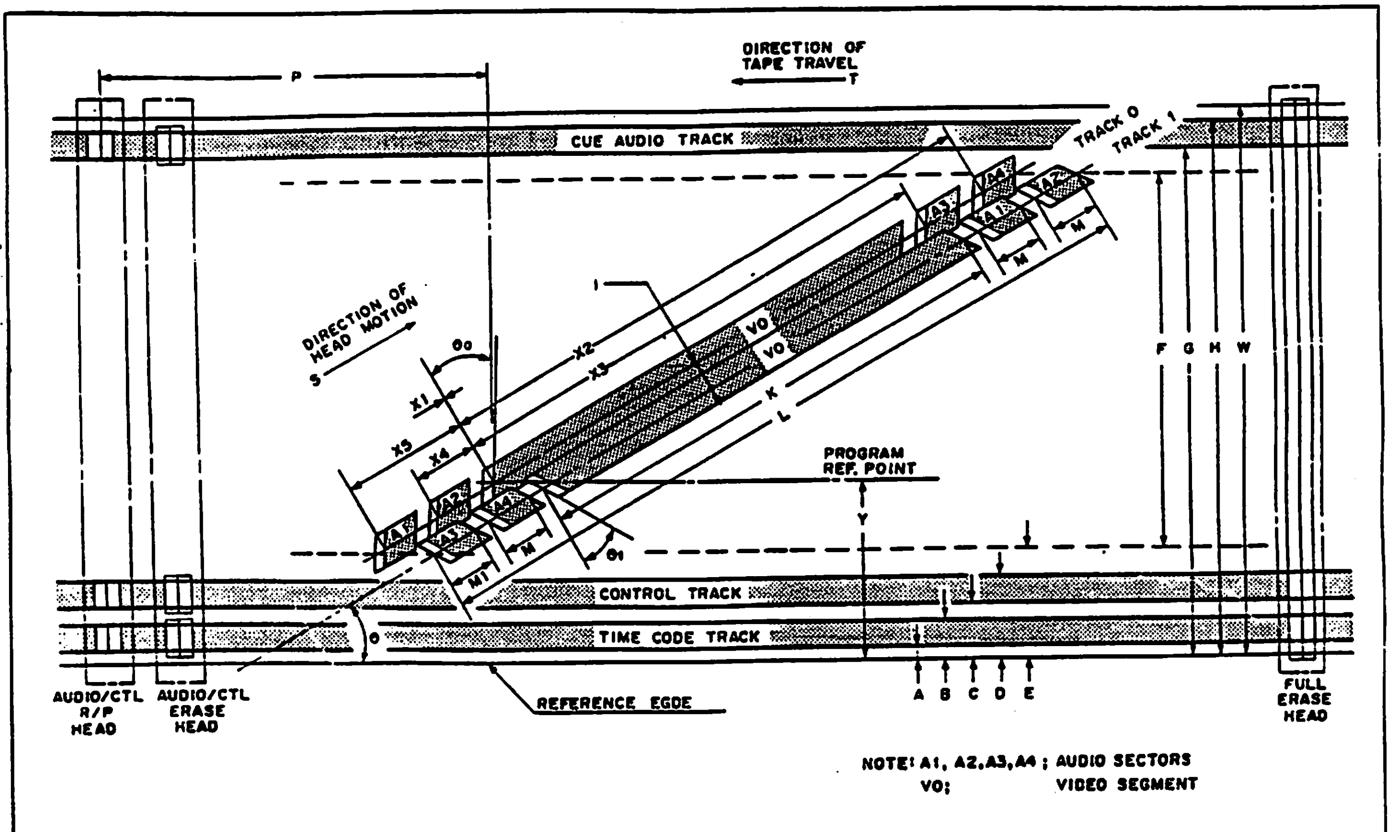


Figure 1-6

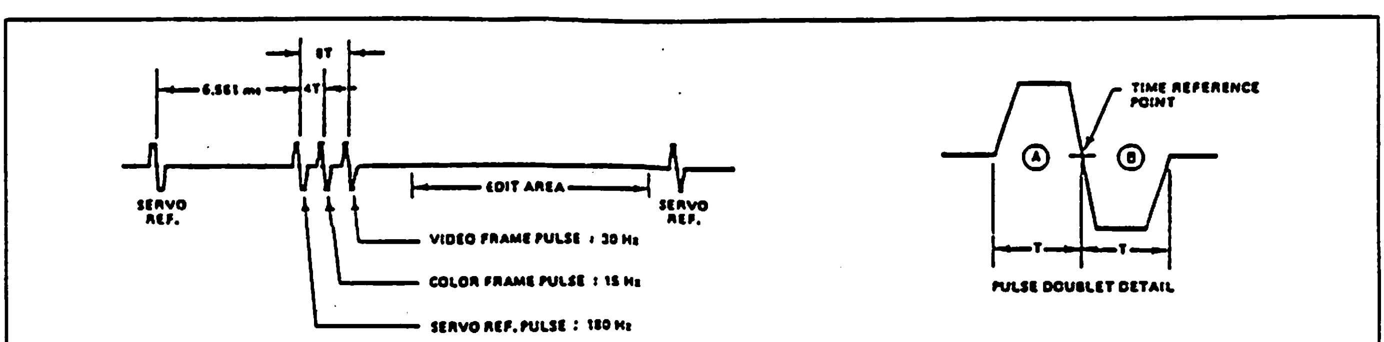


Figure 1-7

Time-Base Correctors

Time-base correctors, also known as TBCs, are necessary because of distortions called time-base errors.

What is a time-base error?
Time-base errors happen when a video or audio signal recorded on a magnetic tape is not played back accurately *in time*. Remember that one line of video lasts 63.5 microseconds. If a video track is read off the tape even a few millionths of second inaccurately, the video signal will not be reconstructed accurately. Large time-base errors will result in color information being replayed incorrectly in time (and hue shifts will result) or line/sync information being replayed incorrectly (luminance will play inaccurately or the monitor will not begin scanning lines at the correct place.

The time-base error a viewer will experience is one of a poorly scanned image, delayed color information, or shifted hues. The deck itself doesn't notice that anything is wrong, however. The signal output from the VTR is as good as it gets.

Something to compare to... a reference
A time-base corrector needs a good, strong reference signal to have a comparison for the erroneous signal coming from the VTR. This signal is called the *reference video signal*. The reference video provides a reliable signal for the TBC and the VTR to "lock" its output to. This means a VTR video output with poor synchronization pulses (or other malady) can be corrected by processing it until it matches the reference signal. There are several possible names for the same basic reference information:

Black Burst signal	Contains horizontal (line) and vertical (field) sync pulses (-40 IRE), BLACK (7.5 IRE), and color subcarrier BURST with each sync pulse
Video reference	The sync pulses, color subcarrier bursts and black level for the TBC to reference
Sync	Sync pulses for horizontal and vertical reference

How a TBC works, basically
A time-base corrector digitizes and stores up to one frame at a time in a digital memory, then compares the video signal coming in to the studio reference signal. The TBC can correct the time-base errors by delaying or advancing the current frame by the amount necessary to make it match the studio reference signal. Then the time-base corrected signal is output.

Because the TBC has a frame buffer to correct timing errors, it also has processor amplifier controls, otherwise known as a *proc amp*. A proc amp allows adjustments of video black (or pedestal, setup), video gain (white), chrominance, hue, and in some cases luminance-to-chrominance (Y-C) delay. There are also horizontal timing and subcarrier phase adjustments.

All VTR's must have TBC's!
This is far from true, but in any practical studio configuration, each VTR's output needs time-base correction. For example, in a production/post-production video switcher, every video signal must be entering so that their sync pulses begin at the same time. If not, when attempting to mix between two sources with different sync pulses, the monitor will jump as it locks from one VTR to another.

Locking all of the VTRs' video signals together is a potentially complicated issue. Fortunately, the way technology's advanced, almost all professional VTRs have TBC's. The necessary adjustments can be made from the deck itself or from a REMOTE TBC connection.

Locking all decks together means sending the same video reference to all of the video decks. This is done with the *house sync generator*. Other names are:

Black burst generator
Master clock
Synchronization
House reference
House sync

Whatever you call it, it provides the same, common video reference to all of the VTR's, edit controllers, time code generators, etc. It is the device that truly designates when all of the VTRs' sync pulses begin and end.

A device without a TBC will entering the system nonsynchronously, and it will never accurately reproduce the original signal.

Editing

Once you have collected all of your footage and the production phase is complete, you are ready to edit. The post-production phase begins...

Editing means selecting specific portions of your original footage and organizing it in a particular way... a way that communicates your intent or message.

Film

Editing video tapes is not at all like editing rolls of film, though it used to be. Motion picture is a series of still images that are moved through a projector 24 times every second. Each still image is called a frame. A frame typically represents 1/24th of a second. This means 24 photographs were taken in one second, and 24 more the next second, and on and on.

Cutting between two of those frames produces a *cut*. Attaching another piece of film at the cut point is called a *splice*. Now two pieces of film are connected and when they play back through the projector, the pictures will shift according to the splice. The process of cutting and splicing all of your footage together in an organized way that communicates your ideas is called editing. Remember that this process has only described the VISUAL portion of the editing... not the AUDIO.

Video

Early videotape editing was done in a similar manner: cut between frames and then splice two pieces of video tape together producing a cut between two images. On film, this is trivial because you can actually see the two frames and the space between them. If you have ever looked at the magnetic tape that stores video information, you will know that it *looks* blank. The magnetic alignment of particles is not visible to the human eye. To see how the magnetic particles on a tape are aligned, special magnetic ink is placed on the tape. This would allow you to see the individual tracks *if* your eyes were small enough to distinguish the tiny tracks. Because video tracks are so small (0.13mm for a1" video tape), it's necessary to use a microscope to distinguish between individual tracks on the tape which are revealed by the magnetic ink. Cut along a specific video track with a razor blade, splice this tape to another tape trying to match track lines, and the edit is made.

Another way to edit Film

Imagine another film editing system: for each shot you would like to edit together, you project light through the *source* film and allow it onto a second, new piece of film (the *master*). For the next shot you would project light from another piece of source film to the master film. When you were done projecting different source film onto the master film, you would develop the master roll of film and project it. What you would see is each source shot, one after another in time, as you had laid them down onto the master film. This is the basis of how an *optical printer* works, because you are optically transferring information from one roll of film to another roll of film.

Creative, Selective Dubbing
To transfer information from one roll of film to another requires an optical transfer because the chemical emulsion on the surface is the film is photosensitive (sensitive to light). To transfer the magnetic tracks of information from a source video tape to a master tape requires sending an electrical signal from the source tape deck to the record deck, where it will be converted back into magnetic information for storage on the master tape. In essence, this form of editing is copying one video tape to another, but only selected parts. Editing is thus creative, selective dubbing.

Clients who request a BetacamSP tape transferred to a VHS tape are asking for a dub.

Clients who request the first five minutes of a BetacamSP tape, then two minutes in the middle, and finally six minutes near the end, and "oh! bars and tone at the head" to a VHS tape are asking for a small edit session.

Online Editing
Any edit session which produces a completed, ready-for-distribution-or-screening video tape is considered an online edit session. Early video edit sessions were always considered online edit sessions.

Meanwhile, in the world of filmmaking:

After a roll of film is shot, it is developed and a *work print* is produced. The work print will be used to make all the creative editing decisions. It will be shuttled back and forth on the film editing reels hundreds of times by the editor. It won't be considered to have high image quality. Its purpose is to refer back to the original, pristine film.

In order for the work print to be referenced back to the original film negative, a special code must be placed on the negative AND on the work print. This code is placed on the edge of the film and is thus called *edge code*. Reading the edge code numbers on the work print will tell the editor what edge code number to look for on the original negative. Once all of the creative decisions about editing are made and the work print is spliced together, the editor will write down all of the edge code numbers and search through the high quality negative for the same numbers.

This results in a two step editing process: offline (the workprint editing stage) and online (the conforming and cutting of the original negative based on edge code numbers).

Dubbing your footage is a good safety measure. Copying your original BetacamSP footage to 3/4" and then editing the 3/4" tape will allow you to put the 3/4" tape through the stresses of fast-forward and rewind and not your original BetacamSP. Dubbing your footage to 3/4" and VHS would allow for a 3/4" safety copy and the VHS would be used to do the creative editing.

The challenge becomes how to reference a video tape dub to a video tape original. Film has edge code numbers. The necessary invention was called *time code*.

Window dubs
The way to reference your inexepensive, off-line VHS or 3/4" dubs is to transfer the time code information to them during the dub.

There are several options:

-Dub longitudinal time code (LTC) to VHS or 3/4" off-line dub, onto an audio track (typically audio track 2)
-Dub vertical interval time code (VITC) from the source tape
-Make a *window dub* of the source tape

The first two options require that a time code signal is transferred to the VHS or 3/4" deck. It also means the VHS or 3/4" deck must have the capability to playback the time code signal (either from audio track 2 or from the VITC)/.

A window dub means that there is a pictoral representation of the time code information (a window) burned over the original image. In freeze frame or when playing the window dub, you are able to see what time code number the original source tape has for that particular frame. Note that just because you see the time code numbers on the picture does not mean there is time code information written on the tape. In other words, a window dub does not necessarily have LTC or VITC.

Edit Decision Lists (EDL's)
The VHS or 3/4" window dubs can now be used to make a creative edit. Each cut laid down will represent the in-point and out-point of the source tape, noted by the time code in the window.

When all editing is done, each in-point and out-point is written down as a separate event. An *edit decision list* is produced from this creative, window dub edit session.

An Edit Decision List (EDL) consists of this information:

Edit #	Transition	Channels	Source In	Source Out	Record In	Record Out
any number	**C**(ut) **D**(issolve) **W**(ipe)	**V**(ideo) **A**(udio) **1** **A**(udio) **2**	Source tape in-point	Source tape out-point	Record tape in-point	Record tape out-point

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SHOW: ____________ PAGE ______ OF ______

EVENT#	TYPE CUT	SPLIT	A/V	SOURCE IN	SOURCE OUT	DESCRIPTION	COMM./REC.IN-OUT	✓
		: :		: : :	: :			
		: :		: : :	: :			
		: :		: : :	: :			
		: :		: : :	: :			
		: :		: : :	: :			
		: :		: : :	: :			
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TWO STEP EDITING PROCESS

The two step editing process is more commonly known as "off-line and on-line" editing. Since the most time consuming element of the video editing process is decision making, an off-line edit is the cut in which all the decisions are made. The off-line is usually done on a system that requires minimal expense (often times in a "cuts only" suite). The on-line is a frame by frame recreation of the off-line cut. On-lines are generally made on considerably more expensive, top quality systems which have both sound and picture effects capabilities.

The two step editing process was developed with the advent of SMPTE time code. Because time code identifies and labels each frame as its own unique frame, accurate recreation of cuts as well as special effects become possible. A project that will be cut in the two step process requires time code on the original (camera shot) footage. Ideally, the original footage will be shot on equipment that records time code along with the picture and/or sound. However, time code can be added after the shoot by "post coding", which is recording the time code on an available audio channel. Copies of the original footage in which the time code numbers are character generated on to the picture are then made. These copies are called Window Dubs and are the source material used in the off-line edit. Because the time code numbers are visible on the screen, a time code editing system is not required for the off-line, allowing for the least possible expense in the off-line step. Once all edit decisions are made, an "edit decision list" is used to edit the original (camera shot) footage in the on-line. Since an on-line suite is time code based, the visual effects such as dissolves and wipes, digital effects, and audio sweetening are all added in the on-line step.

In this two step editing process an accurate edit decision list (also called an EDL) is crucial. EDLs can be either hand or computer generated. When the off-line is cut on the most

simple "cuts only" non-time code system, the list must be done by hand. A hand generated EDL is created by writing down the number of the first and last frame of each shot, the type of cut (video, audio channel one and/or two), and any comments necessary. This is done easily since all time code numbers are visible as part of the picture in a window dub. However, a problem arises in audio only cuts since there is no visual in which time code numbers can be seen. The best solution to this problem is to write down the numbers on any cut that might be covered with other picture at the time the cut is made. On occasion however, the editor might need to look back at un-cut window dubs to correctly identify the exact frame an audio cut starts or ends on. A hand generated list can be entered into a computer program such as "Edit Lister" which can then be loaded into the on-line edit system's controller (computer), allowing the on-line cut to go even faster. When the off-line is cut on a time code system it is possible to attach a computer with the "Edit Lister" program so that each edit's information outputs to the computer. This creates a computer generated EDL which can then be loaded into the on-line system. Since edits are frequently redone and reworked in an off-line, the computer generated list (which lists <u>every</u> edit) always requires some editing of its own. This process is known as "list management" and is done to the off-line list before it can be entered into the on-line computer's system. Some on-line controllers have an Auto Assembly capability in which the pre-loaded list is assembled by the controller with humans only changing tapes at the appropriate time. As one might imagine this speeds up the editing process considerably, however, it is very rare that an Auto Assembly is not stopped for color correction, audio correction, adjustment of effects and/or a myriad of other reasons.

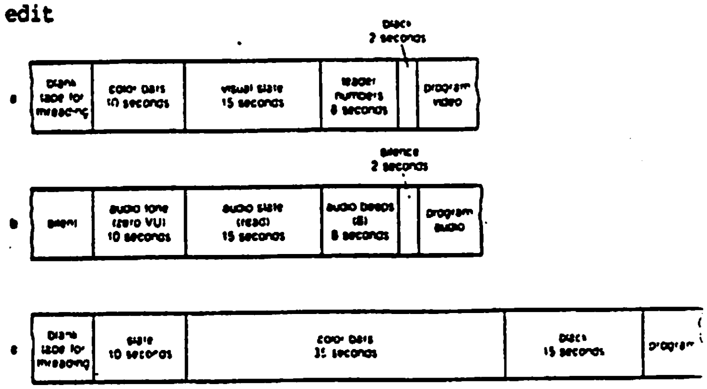
The two step editing process is advantageous in that it allows the user to take advantage of all the possible effects in the most cost efficent way. For example, if a project was only cut in an on-line suite, all the time consuming decisions would also have to be made on that system which could cost up to ten times more than the off-line suite. Additionally, the two step process encourages the editor to take their time with the cut and to rework it until it is the best it can be.

19, KSS

CONTROL TRACK EDITING SKILLS

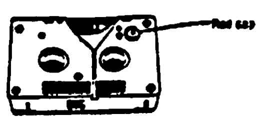
Be sure you get hands-on practice of the following: 171

- __ Turning the system on
- __ Loading tapes
- __ Searching source and master tapes
- __ Selecting edit mode
- __ Entering IN and OUT points for both tapes
- __ Previewing an edit
- __ Trimming IN and OUT points for both tapes
- __ Resetting IN and OUT points for both tapes
- __ Adjusting audio levels
- __ Performing an edit
- __ Reviewing an edit
- __ Checking the edit points
- __ Performing a video-only insert
- __ Performing an audio-only insert
- __ Swithching audio input leads on the record deck
- __ Mixing sound by using two audio channels
- __ Fading audio in and out
- __ Laying a video signal onto a blank tape so you can begin editing in the assemble mode.
- __ Blacking a tape
- __ Performing an assemble edit
- __ Turning the system off



SMPTE video leader (a) and audio leader (b); industrial leader (c)

Safety cap on the bottom
Video cassette incorporates a convenient safety device to prevent accidental erasure.
To safeguard the material recorded on a cassette, remove the red cap on the bottom of the cassette. To record on this cassette later, replace the cap.



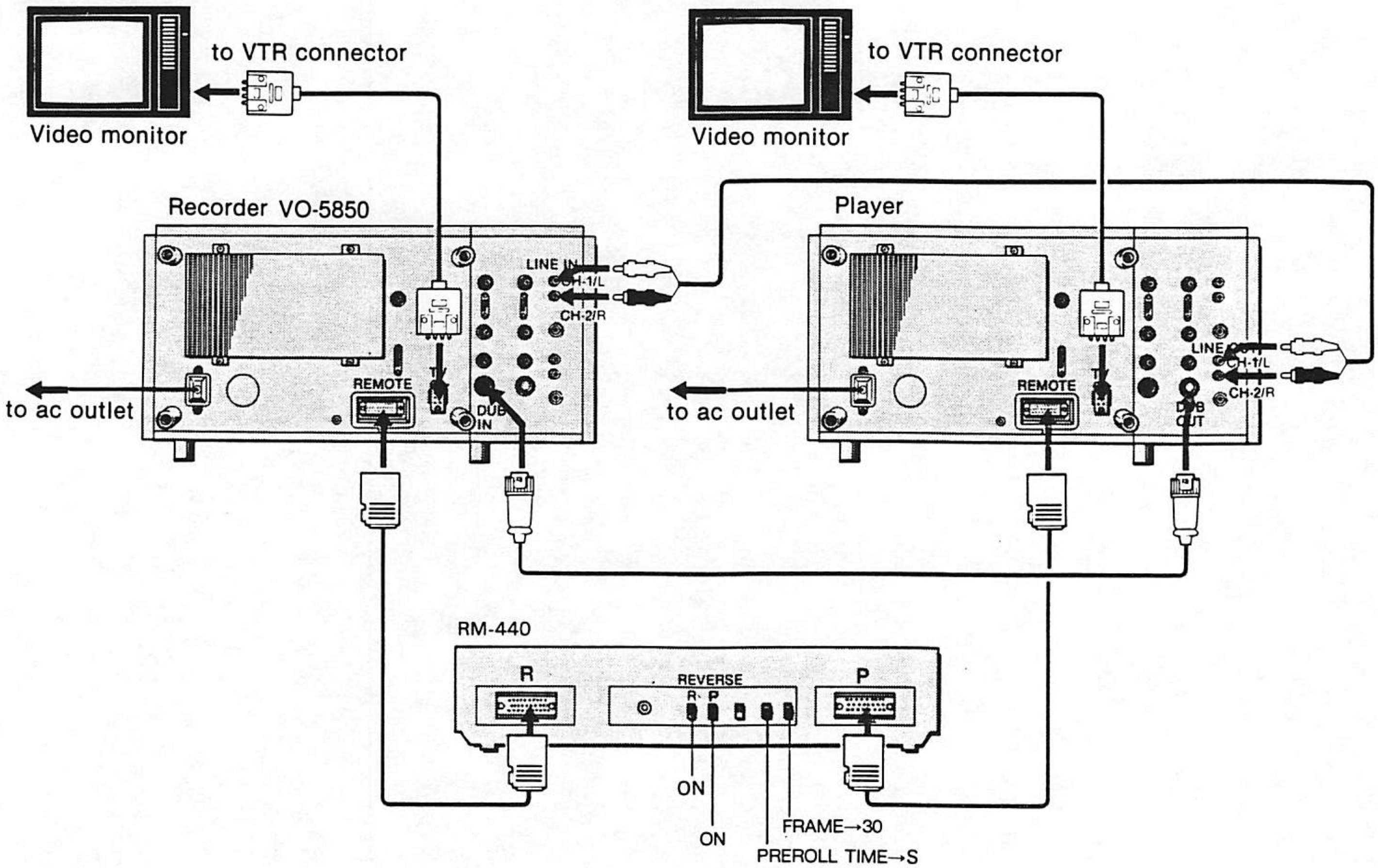
AUTOMATIC EDITING—using the RM-440

When the RM-440 Automatic Editing Control Unit is used with the VO-5850, accurate and automatic tape-to-tape editing is possible. Once the edit-in and edit-out points have been memorized on the RM-440, the editing can be done simply by pressing the AUTO EDIT button.

Almost all functions of the RM-440 can be activated when the RM-440 is used with the VO-5850. However, picture search is possible only when a KCS cassette is used.

● During picture search, the picture may become monochrome or roll vertically on some monitors. If this happens, release the picture search mode by turning the search dial.

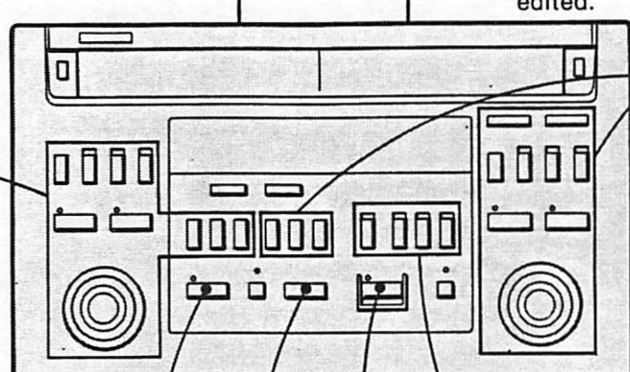
CONNECTIONS



OPERATION

1 Make the preparations on the player.
1. Turn the power on and insert the recorded video cassette.
2. Set the AUDIO MONITOR switch to MIX in most cases.
3. Set the MODE SELECT switch to NORMAL.
(When a time base corrector is used, set the switch to TBC.)
4. Set the MEMORY switch to OFF.
5. Play the tape, and adjust the TRACKING control and SKEW control, if necessary, so that the best possible playback picture is obtained.

2 Make the preparations on the recorder.
1. Turn the power on and insert a video cassette.
●Check that the red button on the bottom of the cassette is in place. If it is not, the E-to-E mode picture will be cut off.
2. Set the AUDIO MONITOR switch to MIX in most cases.
3. Set the MODE SELECT switch to EDIT.
4. Set the INPUT SELECT switch to DUB.
(If the player is connected to the VIDEO IN connector, set the INPUT SELECT switch to LINE.)
5. Set the MEMORY switch to OFF.
6. Play back the tape on the player, and adjust the video recording level if the video signal is to be edited, or the audio recording level if the audio signal is to be edited.



4 Search for the edit-in and edit-out* points on the player and memory them.

5 Search for the edit-in point on the recorder and memory it.

6 Press the PREVIEW button to rehearse the editing.
If necessary, modify the editing points with the TRIM buttons.

3 Decide the edit mode, assembly editing or insert editing.

7 Press the AUTO EDIT button, and the editing is performed automatically.

To check the edited result, press the REVIEW button.

* The edit-out point can be decided on the recorder instead of on the player.

● For details on the procedure of editing, refer to the instruction manual furnished with the RM-440.

LOCATION OF PARTS AND CONTROLS

The following explanation can be used as an index, for the pages on which the parts and/or controls are used, are shown in parentheses.

ENTRY button for the recorder
To enter the edit-in and edit-out points for the recorder, use these buttons (page 11).

TRIM buttons
The memorized edit-in and edit-out points can be modified frame by frame with these buttons (page 17).

ENTRY buttons for the player
To enter the edit-in and edit-out points for the player, use these buttons (page 10).

Digital Time Counter
Counts the CTL pulses on the tape and displays the results in hours, minutes, seconds and frames (page 20).

The function of these controls is the same as those on the player.

- STOP button
- REW (Rewind) button
- PLAY button
- F FWD (Fast Forward) Button
- PAUSE button

SEARCH button
(page 20)

ASSEMBLE button
Press this button for assembly editing (page 8).

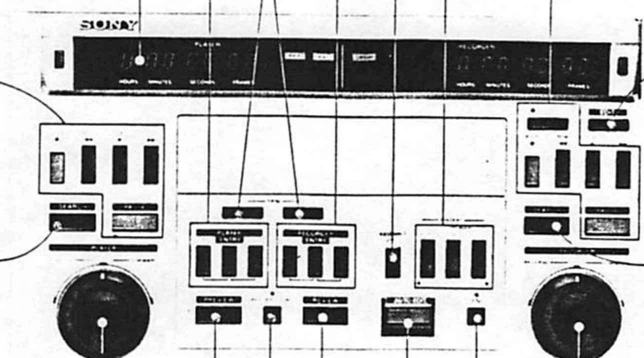
INSERT button
These buttons select the input signal during insert editing (page 8).

The function of these controls is the same as those on the recorder.

- REC (Record) button
- STOP button
- REW (Rewind) button
- PLAY button
- F FWD (Fast Forward) button
- PAUSE button

EDIT button
To do the manual editing, press this button and the PLAY button simultaneously (page 19).
When only this button is pressed, the picture from the player will be monitored on the recorder monitor.

SEARCH button (page 20)



Search dial
The playback speed will be changed from slow to still and to high with this dial (page 10).

Search dial (page 10)

PREVIEW button
This button is used for editing rehearsal. Picture to be edited can be monitored before actual editing (page 14).

RETURN/JUMP button
During the preview mode, press this button, and the recorder will return to the edit-in point at high speed and the still picture can be monitored (page 14).
During the review mode, press this button, and the edited picture between several seconds prior to the edit-in point and 2 seconds after the edit-out point can be monitored (page 15 and 21).

END button
During the manual editing, press this button, and the editing will be cut out (page 19).

During the preview mode or the automatic editing, press this button, and that point will be the edit-out point (page 13, 16 and 18).

AUTO EDIT button
Press this button to put the recorder in the automatic editing mode (page 12).

REVIEW button
Press the button to review the edited picture (page 15).

CONTROL TRACK and TIME CODE

CONTROL TRACK is a synchronizing pulse like an orchestra conductor's baton. Control Track refers to synchronizing pulses recorded on a portion of the videotape called the control track. There is essentially a pulse for each frame of video — 30 per second. These pulses are used by videotape recorders to keep the tape moving at exactly the right speed and position so that the playback head reads the video signal on the tape just as it is supposed to appear on the screen of the TV. Without these "synch" pulses, as they are sometimes called, the VTR would be unable to play the videotape accurately, and the signal appearing on the TV screen would be jumbled. A blank tape has no control track. Whenever a camera records an image, it is also laying down control track. Whenever a videotape is played back, a VTR is using the control track pulses from that recording to "read" the information correctly and form frames of picture and sound. Stable image requires stable control track.

TIME CODE can be recorded in addition to the video, audio and control tracks. It does NOT replace control track. Time code is an exact address, as on a mailbox, for each frame of video. It is a computer code recorded on the address track or on one of the audio channels, giving a distinct number, or address, to each frame of video. This address is given as the number of Hours: Minutes: Seconds: Frames from the beginning of the videotape. (The beginning number may be set at 00:00:00:00 by the user, or any other useful number. Since most tapes shot in the field are less than one hour, the hour section of the time code number is often preset to indicate the reel number.)

EDITING VIDEO

SOURCE: videotapes you are editing from (original tapes, dubs of originals)
MASTER: the videotape you are editing onto (the final, completed program)
IN: an edit start point, can refer to either the beginning of selected material on the source tape or the point on the master tape where that material will begin recording.
OUT: an edit end point on either the source tape or the master tape.

Editing film and video are different processes. Film is physically cut and spliced together. You would never cut videotape except in unusual circumstances. A splice traveling past the heads of a deck (video tape recorder) could damage them. VIDEO DECKS ELECTRONICALLY COPY VIDEO PICTURES AND SOUND FROM SOURCE TAPES TO MASTER TAPE. Audio and video on the original are copied and not erased. The finished edited tape is an edited master.

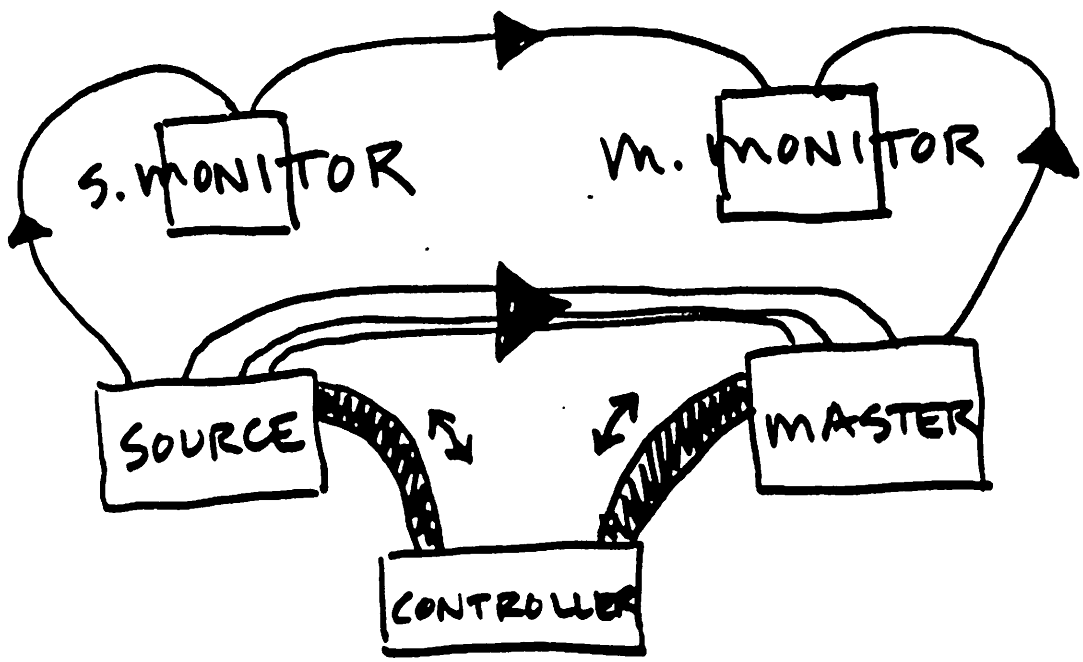
Control Track vs Time Code Editing

All video editing is done with an editing controller that is a computer specially designed to control videotape players and recorders so that video and audio segments may be transferred from one tape to another with great precision. The editing controller may use control track pulses or time code numbers as a reference for making each edit occur exactly where the user indicated it should occur. Time code editing is more accurate because the computer knows the exact address of the desired edit point and can find it again, without fail, by reading the time code.
CONTROL TRACK EDIT CONTROLLERS USE CONTROL TRACK PULSES AS A REFERENCE TO BE ABLE TO LOCATE A DESIRED EDIT POINT AFTER THE VIDEOTAPE HAS BEEN MOVED AWAY FROM THAT POINT. IT DOES THIS BY KEEPING COUNT OF CONTROL TRACK PULSES AS THE TAPE MOVES. Control track editing is less accurate: sometimes the computer skips a pulse or two. And, if you remove the tape from the machine, the computer loses reference and counts from wherever it finds itself. Pressing STOP may cause the counter to skip or gain a few frames once the tape is brought back to the heads.

Basic Equipment Configuration

A basic edit suite will have:

Source deck........................	Plays back source tapes.
Source monitor....................	Displays signal from source deck.
Record deck........................	Records signal from source deck onto master tape.
Record monitor...................	Displays signal being recorded or already recorded on master tape.
Controller............................	The central, computerized brain of the system; controls all deck functions and edit functions.



Controllers

Many basic controllers have three groupings of controls: left, center and right.

Deck Functions

The left group of controls operates the source deck and the right group of controls operates the master deck. There is a complete set of STOP, REWIND, PLAY, FFWD, PAUSE and SEARCH buttons on each side. Use these rather than the controls on the deck itself.
The LED Counter Number Display indicates, in hours, minutes, seconds and frames, the current position of each tape. You can reset them to zero at anytime, using the reset buttons. Notice the frame readout runs from 1 to 29 and then starts at 1 again. This is because in video there are 29.97 frames per second.

Edit Functions

ENTRY IN and OUT: Press ENTRYand IN to mark the point on the tape where the edit should begin. Press ENTRY and OUT to mark the point on the tape where the edit should end. These selections can be chosen for both source and master tapes. The controller will accept up to three entries, but only requires two entry points to complete an edit.
TRIM+/- : Adds or subtracts frames at IN or OUT points. Hold down selected IN or OUT button and then press (+) or (-) once for each frame to be trimmed. The Counter Number Display will advance or back up accordingly.
PREVIEW: Allows you to see the edit in real time, without actually recording it onto your master.
PERFORM: Actually records the selected segment onto the master tape.

How an Edit Occurs

The IN and OUT points of edits are marked and memorized using the edit controller. When the controller previews or performs the edit, it will:

1. Roll both source and master tapes back several seconds (often, 3-7 seconds) BEFORE the selected IN points. This is PRE-ROLL.
2. Plays both tapes, using the pre-roll time to synchronize the tapes' control tracks.
3. Transfers the source material to the master once the tapes reach the IN points.
4. Stops transferring source material once the OUT points are reached.
5. Plays both source and master tapes several seconds beyond the OUT points. This is POST-ROLL.

Generations

The source footage are first generation tapes. The master tape is second generation because it is copied from the source tapes. A copy of the master is third generation and so on... As the number of generations is increased, video signal quality degrades. Dubbing or editing through signal processing equipment such as a Time Base Corrector(TBC) will result in cleaner copies and a Processing Amplifier(Proc Amp) allows for a certain amount of color and light correction.

EDIT MODES - ASSEMBLE and INSERT EDITING

Edit Mode Selection

There are two modes of editing available to choose from on any controller: ASSEMBLE and INSERT. Both modes require several seconds of stable control track (i.e. any recorded image) PRECEDING the desired edit points, so the system can pre-roll, syncing up the control tracks of the two tapes for a clean edit.

Assemble Edit

An assemble edit always records audio, video and control track; the tracks cannot be laid down independent from each other in this mode. An assemble edit is "clean" (stable) at the in point, but "dirty" (image rolls or breaks up) at the out point. This mode always breaks control track at the end of an edit, which destabilizes the image at that point, causing the "dirty" out. Assemble editing is only useful, generally, to piece chunks together quickly, without separating tracks nor replacing any previous edits.

FEATURES

- Always records all tracks: video, audio, control track
- Clean in, DIRTY OUT
- Requires pre-roll
- Can edit onto a blank master tape
- SHOULD NOT BE USED TO EDIT INTO PREVIOUSLY RECORDED MATERIAL
  Control track will be broken in the middle of your master, your baby, your gold, your gem. you don't want that, do you?
- good for slamming chunks together

Insert Edit

Insert editing was developed as a means of editing new material into previously recorded material without having video break-up at the out points. Insert edits produce "clean" ins and outs. This mode also enables editing of video, audio channel 1, or

audio channel 2 separately or in any combination. Insert edits record video and/or audio, but DO NOT record control track. This means that a control track must first be recorded on the master tape.

In order to record control track on your master in preparation for an insert edit, video must also be recorded. The video image of choice is black. Deep rich chocolate velvety black. This is called **laying down black** or blacking your master tape. Now that black, and the control track that came with it, have been recorded onto the master tape, the source and master decks can sync up during pre-roll and the video and audio will transfer to a master that has a stable and continuous control track.

With insert editing, you may edit :

VIDEO ONLY
VIDEO/CHANNEL 1 AUDIO
VIDEO/CHANNEL 2 AUDIO
AUDIO ONLY CHANNEL 1
AUDIO ONLY CHANNEL 2
AUDIO ONLY CHANNEL 1 & 2

FEATURES

- can edit any combination of
  - Audio Channel 1
  - Audio Channel 2
  - Video
- clean in, clean out
- requires pre-roll
- requires a BLACKED master tape
- allows great creativity and flexibility in video editing

Hard Record

A hard edit is simply pressing Play and Record as you would on a home VCR to record a TV show. It requires no controller, no pre-roll, and can be recorded on a blank tape. It has a "dirty" in and a "dirty" out. It's only use for editing is to record from the very head of a tape to the very tail (or well beyond any desired material). This is how you would record black when blacking a master for insert editing.

Blacking

Black and control track can be generated in several ways. Patch any of the following to a deck and hard record this signal onto your master tape. Don't stop or pause the deck; let it record non-stop through the entire tape for a clean, continuous control track.

1. A sync, or black burst, generator. It will provide a stable sync signal as well as a solid black image.

2. A camera, powered up and with the lens cap on. The camera must be on so it is generating a sync signal for the deck to record. The image will be the lens cap.
3. Another deck playing back a tape that is already blacked. If the signal is unstable at any point, your new blacked master will have recorded the same problem.

Always record more control track than you will possibly need. The extra few minutes spent recording the control track will save you much time later, if you find that your show is running longer than expected.

YOU CAN USUALLY PURCHASE A BLACKED MASTER IN THE FORMAT OF YOUR CHOICE AT AN EDITING FACILITY.

COMMON EDITING MISTAKES

1. **If the numbers reading time are not moving** - this means there is no control track on that part of the tape and the controller can not use control track to count. Most commonly, this problem appears when there is not enough stable control track BEFORE the in point, preventing pre-roll.

2. **Setting the same point as both IN and OUT** - Most controllers will not accept the entry or will cancel the first point when the second is entered. You can tell this is the problem if you press the IN and OUT buttons and see the same point is recorded. Reset or cancel the incorrect point.

3. **Setting an OUT point earlier on the tape than the IN point** - The controller may refuse the entry or refuse to PREVIEW. Reset or cancel the incorrect point.

4. **Forgetting to set an IN point** - When you push PREVIEW, the controller will set the IN point where the tape is currently paused.

5. **Trimming the wrong direction or the wrong point** - Sometimes you have to sketch out which direction you are trimming. To include more of the image does NOT necessarily mean adding nor reducing the image necessarily mean subtracting.

6. **Forgetting to set audio levels** - Do not be fooled by the volume you choose on the master monitor; always check the meters on the master deck while previewing. Do not be fooled by listening to the source monitor during the edit; turn it down.

7. **Forgetting to PERFORM the edit** - This will happen more often than you think. If you REVIEW each edit, you'll catch this mistake before losing your IN and OUT points.

8. **Using the wrong edit mode** - Check the edit mode especially when beginning a session. Also check the mode after doing a video or audio insert or a split edit. You should catch this mistake during PREVIEW, but if you selected ASSEMBLE and perform the edit, watch out! You'll either have to continue your basic cuts in the assemble mode or assemble edit black from the "dirty" out to the end of tape in order to continue insert editing.

9. **Controller doesn't accept an edit point or won't preview or perform the edit** - SEE MISTAKES # 1,2, AND 3 ABOVE. Also, one or both of your decks may be stopped and the tape is off the heads.

10. **Edit includes unwanted material** - The system probably slipped a few frames. Check slippage with a practice edit when beginning a session. You may have to reset

edit points after a lot of previewing, or refrain from previewing a scene many times. 2-4 frames is common slippage; you can compensate for this by moving your IN point 2-4 frames in the appropriate direction.

11. **Split edit wasn't split** - You probably punched in the second track during PREVIEW, but failed to de-select the track when you went to PERFORM the edit. These off-line controllers only remember points that are entered, not the point at which you punch in a track.

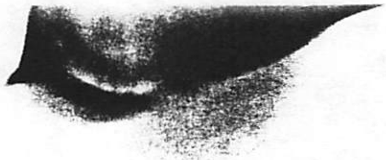
12. **No audio was transferred** - (1) You may have selected the wrong audio channel or no audio channel. (2) Your input leads may be reversed so the audio is going to the wrong channel. (3) You may have forgotten to set levels for that channel. (4) Your monitor volume is turned off. (5) The monitor select switch (on the deck panel) is set to the wrong channel so you can't hear what has been recorded.

OFF-LINE AND ON-LINE EDITING

The most time-consuming element of the video editing process is decision-making. The most wear and tear on original footage is the playing and shuttling of that footage while making those decisions. The off-line/on-line process of editing will save your original footage from abuse and save you money.

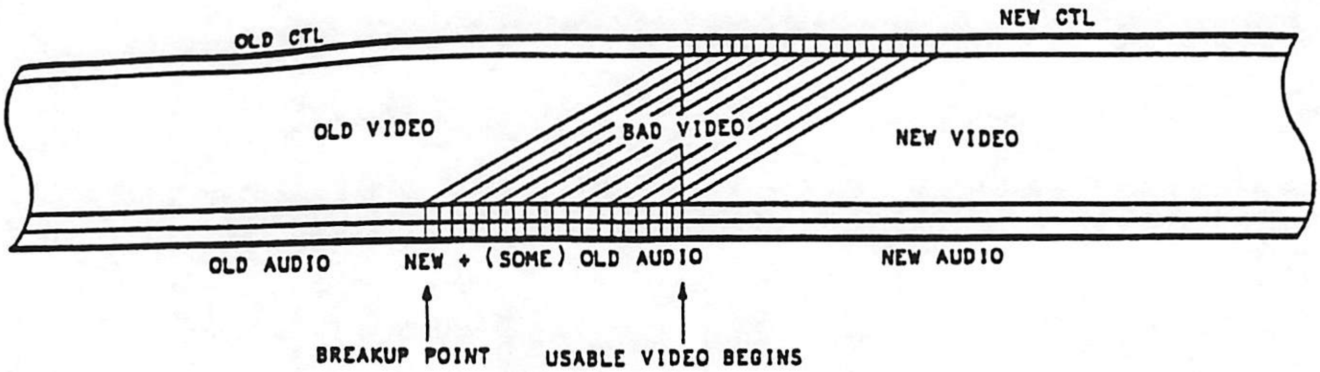
The off-line edit is where all the decisions are made, using DUBS of the original tapes as source tapes. These dubs will suffer the abuse of repeated passes over the heads (shuttling back and forth, playing). You will edit an OFF-LINE MASTER and also keep a list of the edits you make. This list is an Edit Decision List, or EDL. Off-lines are usually done on a system that requires minimal expense, often referred to as a "cuts only" system (commonly, a VHS or 3/4 system). A cuts only system does not have the capacity to perform sophisticated transitions, such as dissolves, It will not generate special effects nor titling, generally, and has little ability to manipulate audio quality (it may include a mixer with basic equalization functions).

The ON-LINE is a re-creation of that off-line master, using the EDL as a guide. The master created here is the actual master, on the best format you can afford or is appropriate, using the ORIGINAL tapes as source tapes. The need to search for edit points or experiment with various shots has been eliminated, since these decisions have been made in the off-line. On-lines are generally made on more expensive, higher format systems with the capacity to do complex transitions, special effects, and some audio sweetening.



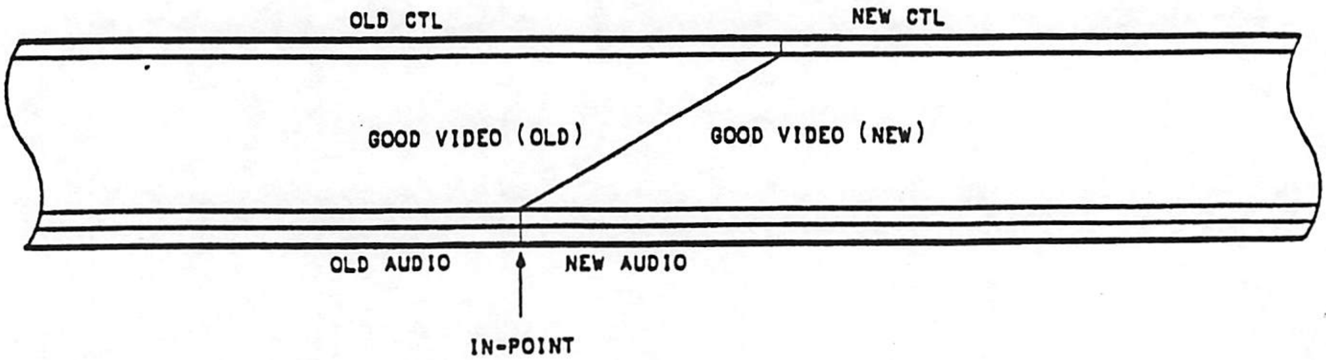
TYPES OF VIDEOTAPE EDITS

CRASH EDIT



ASSEMBLE EDIT

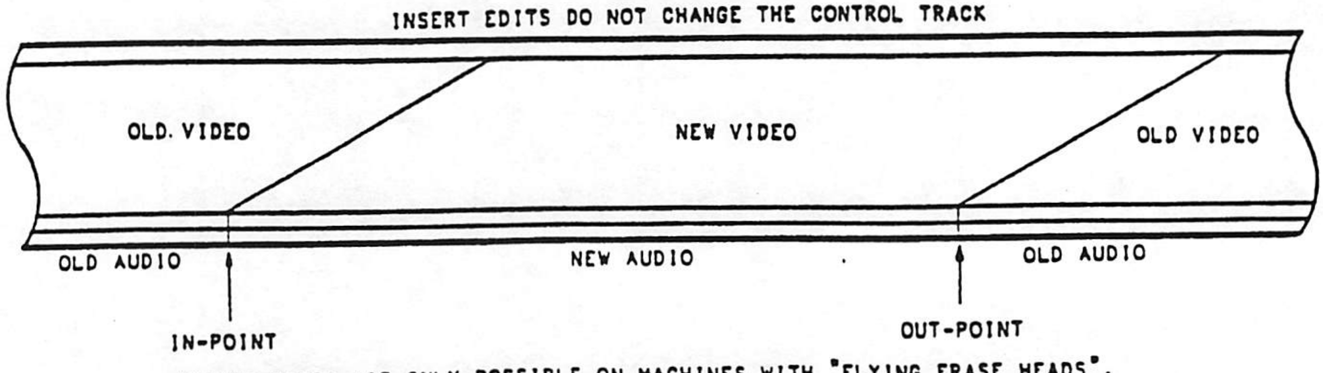
CONTROL TRACK, VIDEO, AND AUDIO ARE ALL EDITED.



ASSEMBLE EDITS ARE ONLY POSSIBLE ON VCR'S WITH "FLYING ERASE HEADS".

INSERT EDIT

INSERT EDITS DO NOT CHANGE THE CONTROL TRACK



INSERT EDITS ARE ONLY POSSIBLE ON MACHINES WITH "FLYING ERASE HEADS".
THE OPERATOR MAY SELECT ANY COMBINATION OF AUDIO 1, AUDIO 2, OR VIDEO

PERFORMING AN EDIT

- Turn the system on
- Check settings on deck panels
- Load your master tape and a selected source tape
- Select EDIT MODE: to insert edit, simply choose the track or tracks you want to edit (V, A1, A2)
- Search source and master tapes
- Enter IN points for both tapes (OUT point optional)
- PREVIEW the edit - check audio levels!!!
- Trim IN and OUT points for both tapes, if needed
- Reset IN and OUT points for both tapes, if needed
- PREVIEW the edit again - check audio levels
- PERFORM the edit
- REVIEW the edit

You may need to:

- Check the edit points
- Switch audio input leads on the master deck
- Fade audio up or down
- Punch in a track selection during an edit (split-edits)
- Black a tape
- Lay a video signal onto a blank tape (enough for pre-roll) so you can begin editing in the assemble mode
- Perform an assemble edit

TYPES OF EDITS

- **STRAIGHT CUT** - Video and audio are cut at the same time.
- **VIDEO Insert** - Only the video is transferred
- **AUDIO Insert** - Only the audio from one or both channels is transferred. It may be faded up or down.
- **SPLIT EDITS (or L-cuts)** - The cut begins with only video or audio being transferred. Later, during the edit, the corresponding audio or video track is punched in.

  VIDEO DELAY - We hear the sound from a scene before we see the video image of it. We may hear someone begin to talk while we are viewing a landscape. Then we cut to the image of the speaker as their talking continues. This softens the transition, seeming smooth and seamless to the viewer.

AUDIO DELAY- We see the image of a scene before we hear the sound. We may see the image of a person talking but hear a narrator's voice introducing them. When the introduction is finished, the voice of that person comes on. Or we may see a waterfall while hearing voice-over. When the voice-over stops, we hear the crashing of the waterfall come in.

- **AUDIO FADE-UPS, FADE-DOWNS** - Gradually turning the audio levels up at the beginning of an edit or down by the end of an edit. Be smooth.
- **CROSS-FADES** - To create a smooth transition between two edits, you can overlap the sounds from the two scenes by placing them on separate audio channels. Fade down the sound at the end of the first edit. For the second edit, fade up its sound DURING the fade-down from the last edit. The result is an audio "dissolve" or cross-fade.
- **MATCH-CUTTING** - matches the action from one image to the action in the next image, such as going from a wide shot to a medium shot of the same action. When match-cutting from close to wide, you might need to overlap the action. When match-cutting from wide to close you might need to underlap the action.

EDITING AUDIO

TYPES OF AUDIO

- SYNCH SOUND - recorded with video at the time of the original recording.
- NARRATION, VOICE-OVER - added to the visuals during editing, usually.
- MUSIC
- SOUND EFFECTS
- AMBIENT SOUND, ROOM TONE

Sound effects can heighten moments, create psychological tension or release, and generally enrich segments of your tape. An abrupt audio transition, such as a straight cut, can emphasize the change to a new scene or event. On the other hand, audio can be used to smoothly join different shots. Often music or ambient sound is laid "under" visuals and voice. Crossfades between ambient sounds and/or musical selections can smooth transitions. Split-edits can ease the viewer into a new scene or setting. Ambiance and room tone can often be used to clean up audio pops and inconsistencies in sound cuts. DESIGNATING PRIMARY AND SECONDARY AUDIO TRACKS DOES NOT MEAN PRIMARY AND SECONDARY AUDIO EDITS WILL ALWAYS BE ON THOSE TRACKS. For instance, cross-fading two musical selections requires you to alternate tracks so a fade-up can occur during a fade-down, resulting in secondary audio on both audio channel 1 & 2.

Discreet Audio Channels

Use only one audio channel for a given sound edit, leaving the other channel blank. This will allow you to put primary audio (voice, for instance) on one track and secondary audio (music, sound effects, or ambiance) on the other. It will also allow you to cross-fade two audio edits. Even if no secondary audio is to be recorded under a given edit, keep the primary audio to one channel. Putting it on both channels will double its volume compared with the single channel audio edits.

Problem Sound, Unwanted Noises

Matching room tone or ambiance can replace specific, short sounds such as barking dogs or sirens that occur in pauses during the primary audio. Simply edit the tone or ambiance onto the portion of the track you wish to replace. If this edit is too apparent, try recording nothing over that portion (turning audio level all the way down, thus recording a silent signal) and record ambiance on the secondary track during that portion, fading it in and out to smooth the transition.

Running audio through a mixer with equalization controls may enable you to minimize the unwanted sound by removing or reducing the frequency range. However, you are also reducing the frequency range of your desirable audio.

These remedies have limited success. More sophisticated processing can be done to clean up your audio, but GOOD AUDIO RECORDING IN THE FIELD IS THE BEST REMEDY.

Room tone or ambiance can be laid under primary audio to imply it was recorded in the same location as other audio or just to locate the primary audio in a specific environment.

Audio Levels

On a standard VU meter, PRIMARY AUDIO levels should be recorded strong, peaking at 0 db or a little above, in the red zone. Set levels on your master deck according to these "peaks", or loudest points during preview. Levels which are too hot (above +3 or the needle "pins" at the end of the scale) risk distorting. Levels peaking too low will sound hissy when the viewer later turns the volume up in order to hear your tape. REMINDER: the human voice and other sounds naturally fluctuate in volume. You do not need to constantly "ride" (change) the levels; generally you will meter for the frequent peaks and let the rest of the audio be. Occasional riding may be necessary for substantial changes in audio levels.

SECONDARY AUDIO levels should be set by ear, in relation to the primary audio. The levels may read low, while sounding perfect with the primary audio. This is normal.

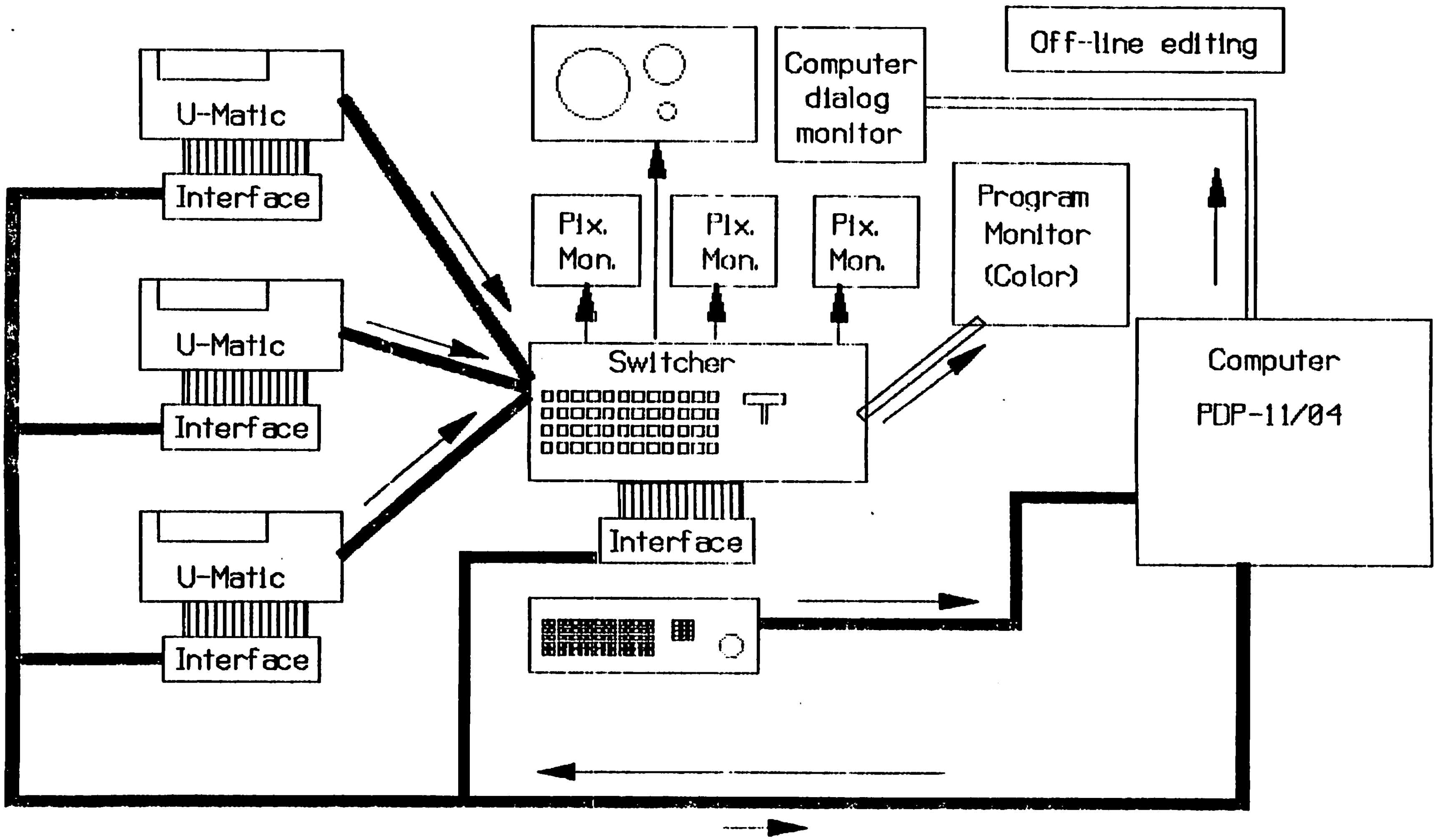
When previewing or performing an edit, turn down the volume on the source monitor, so that you are hearing the master audio recording only. This way you will not be fooled by hearing source audio during the edit.

Voice Over

A microphone can be patched into the master deck and voice-over recorded directly onto your master tape. Beware of extraneous sound in the edit room during recording and monitor recording through headsets. Voice-over can also be recorded in advance to be used as a source tape to edit from.

Planning

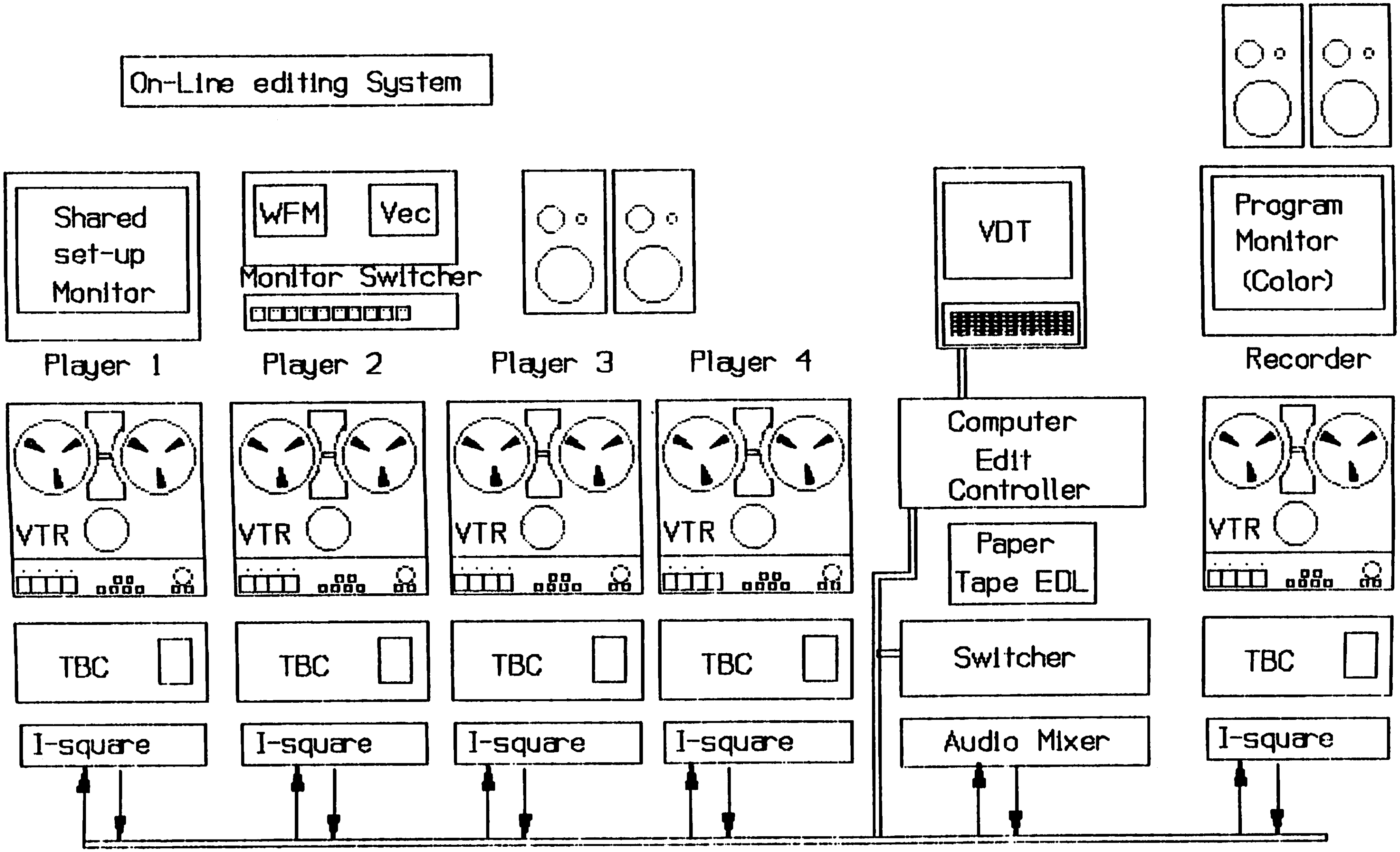
Remember, there are only two audio tracks, so plan which track each audio edit will occupy. Keep track of what has been recorded on each channel while edit so a portion of a previous audio edit does not get erased.
Ambient sound and room tone should be recorded at any location you shoot at. Additional ambient sound and effects can be recorded with a camera and external mic later. Musical selections can be transferred from audio cassette or CD to videotape at an editing facility. Be sure to request in advance that a cassette or CD player be available and configured to dub to your desired video format. A video signal must be recorded with the music to get control track on the tape.



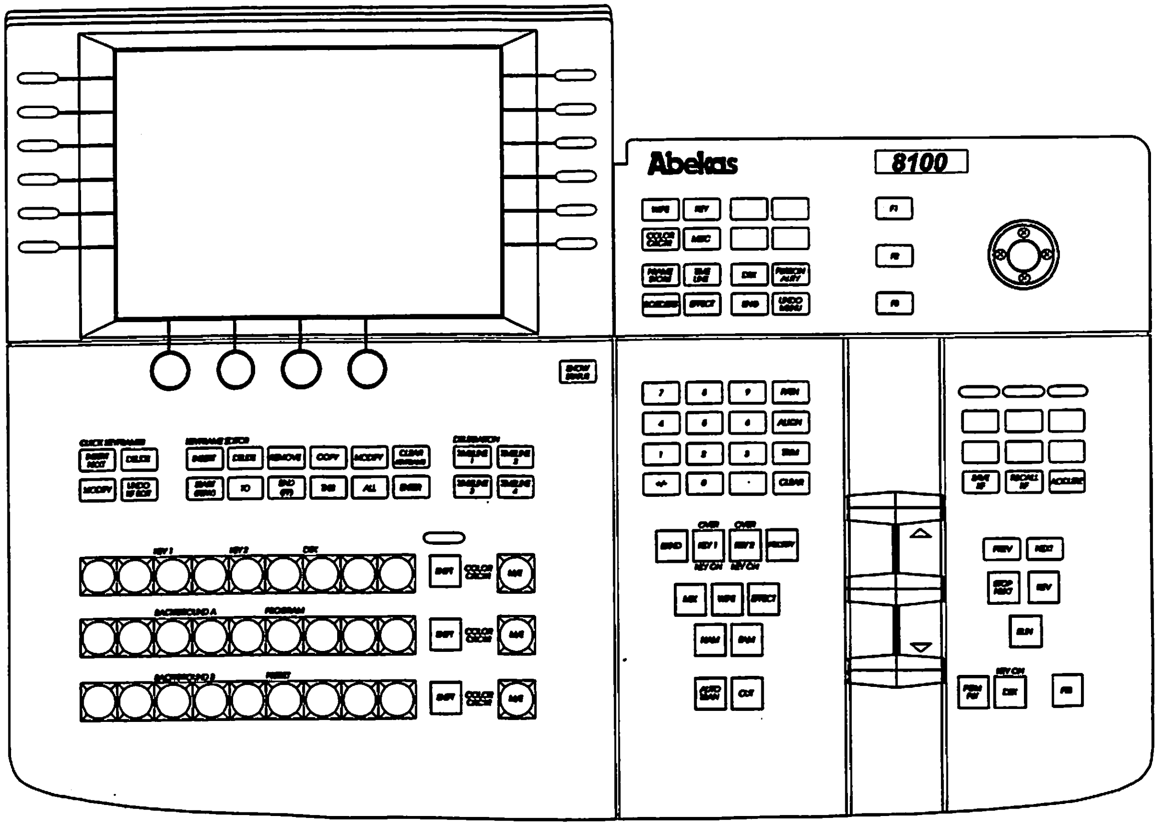
U-Matic
Interface
U-Matic
Interface
U-Matic
Interface
Computer
dialog
monitor
Off-line editing
Pix.
Mon.
Pix.
Mon.
Pix.
Mon.
Program
Monitor
(Color)
Switcher
Interface
Computer
PDP-11/04

185

10


On-Line editing System
Shared set-up Monitor
WFM
Vec
Monitor Switcher
VDT
Program Monitor (Color)
Player 1
Player 2
Player 3
Player 4
Recorder
VTR
VTR
VTR
VTR
Computer Edit Controller
VTR
Paper Tape EDL
TBC
TBC
TBC
TBC
Switcher
TBC
I-square
I-square
I-square
I-square
Audio Mixer
I-square

CONTROL PANEL



SIGNAL CHASSIS

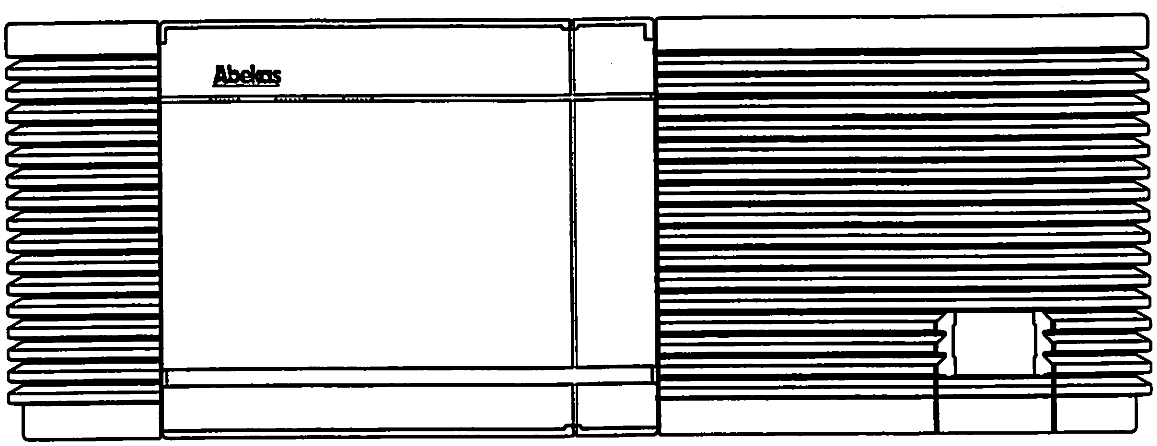


Figure 1-1 8100 System Components

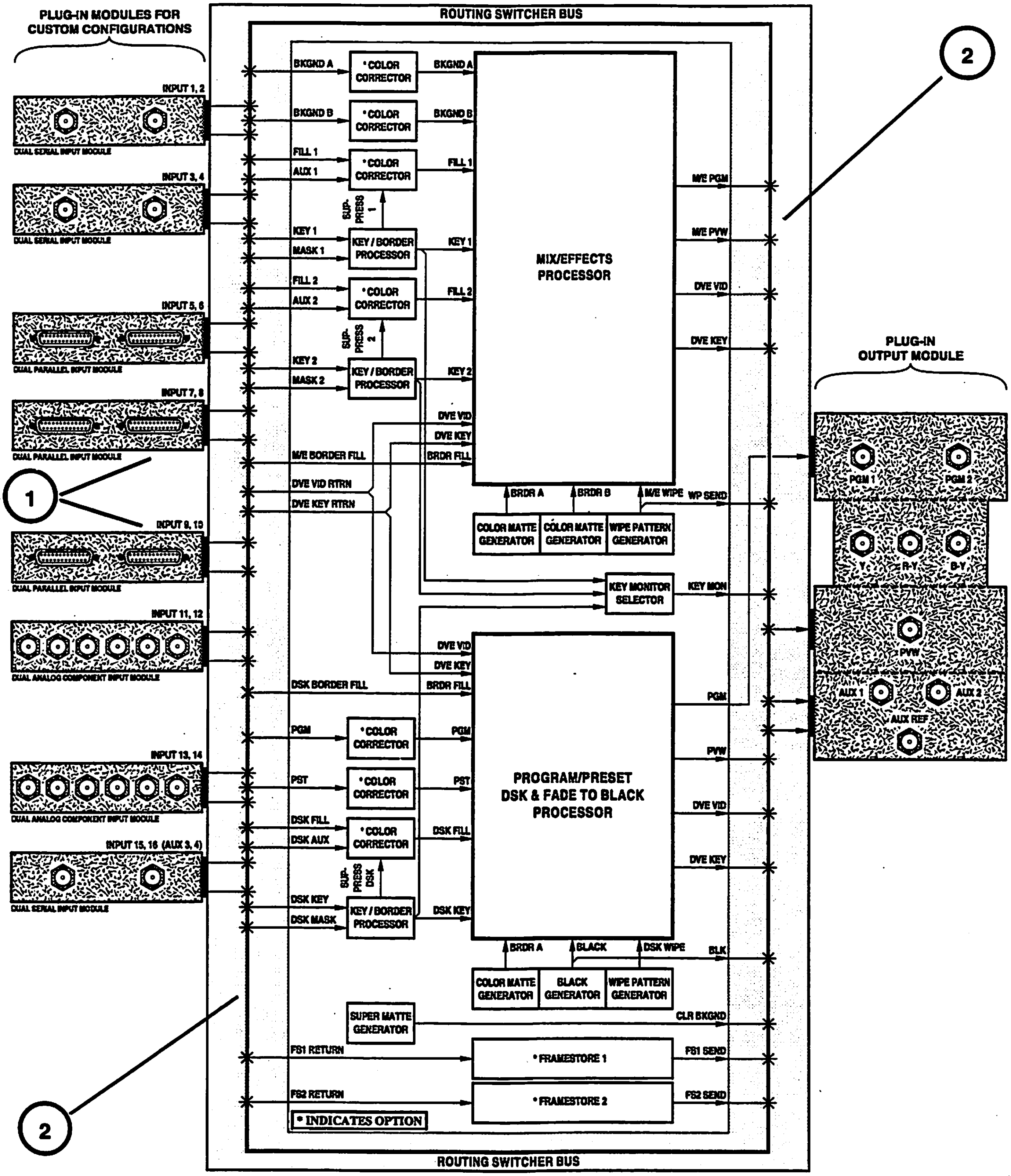


Figure 1-2 8100 Video Functional Diagram

2-1 Construction of Control Panel and Connector Panel

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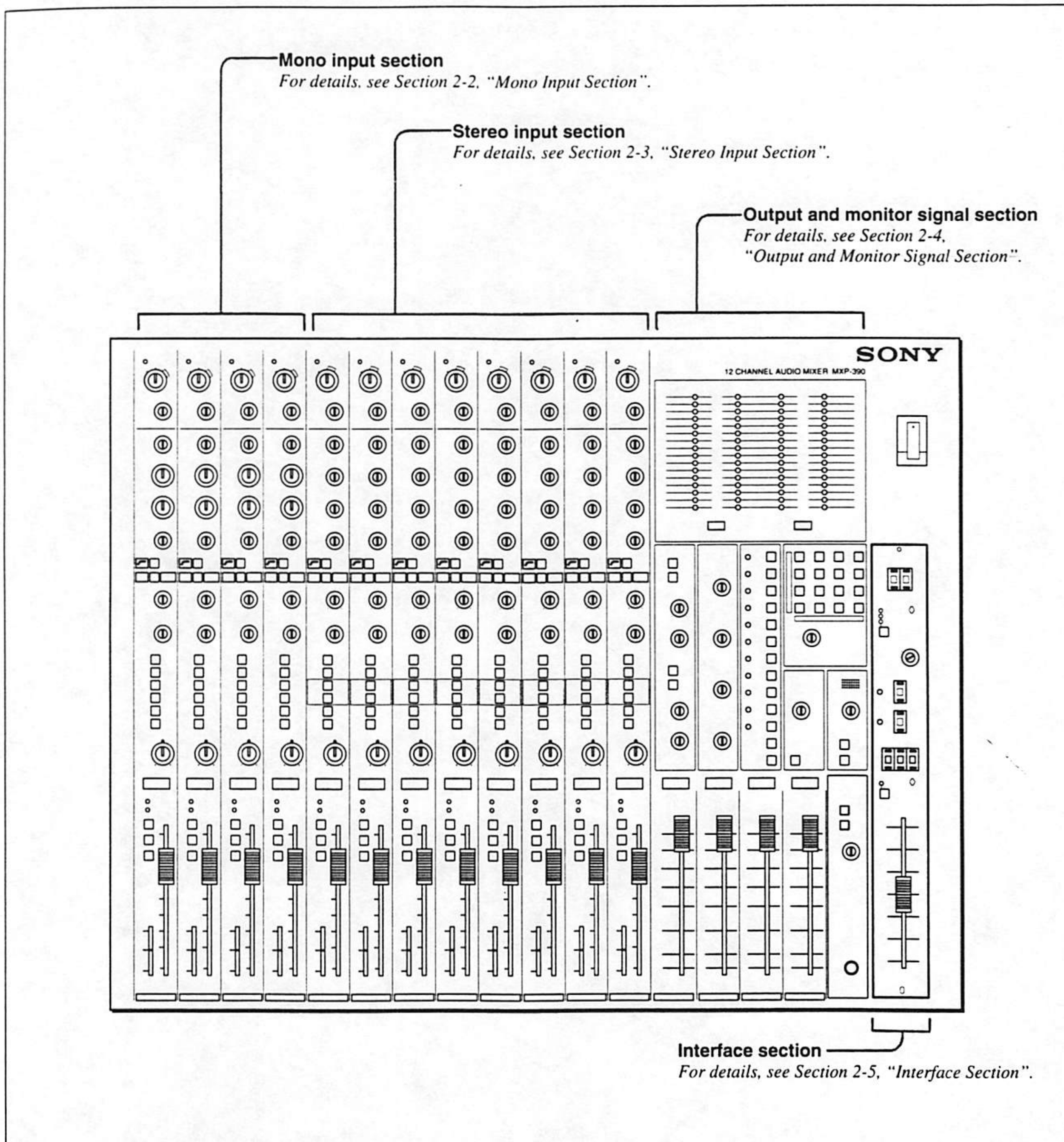
This section shows the construction of the control panel and connector panel of the MXP-390.

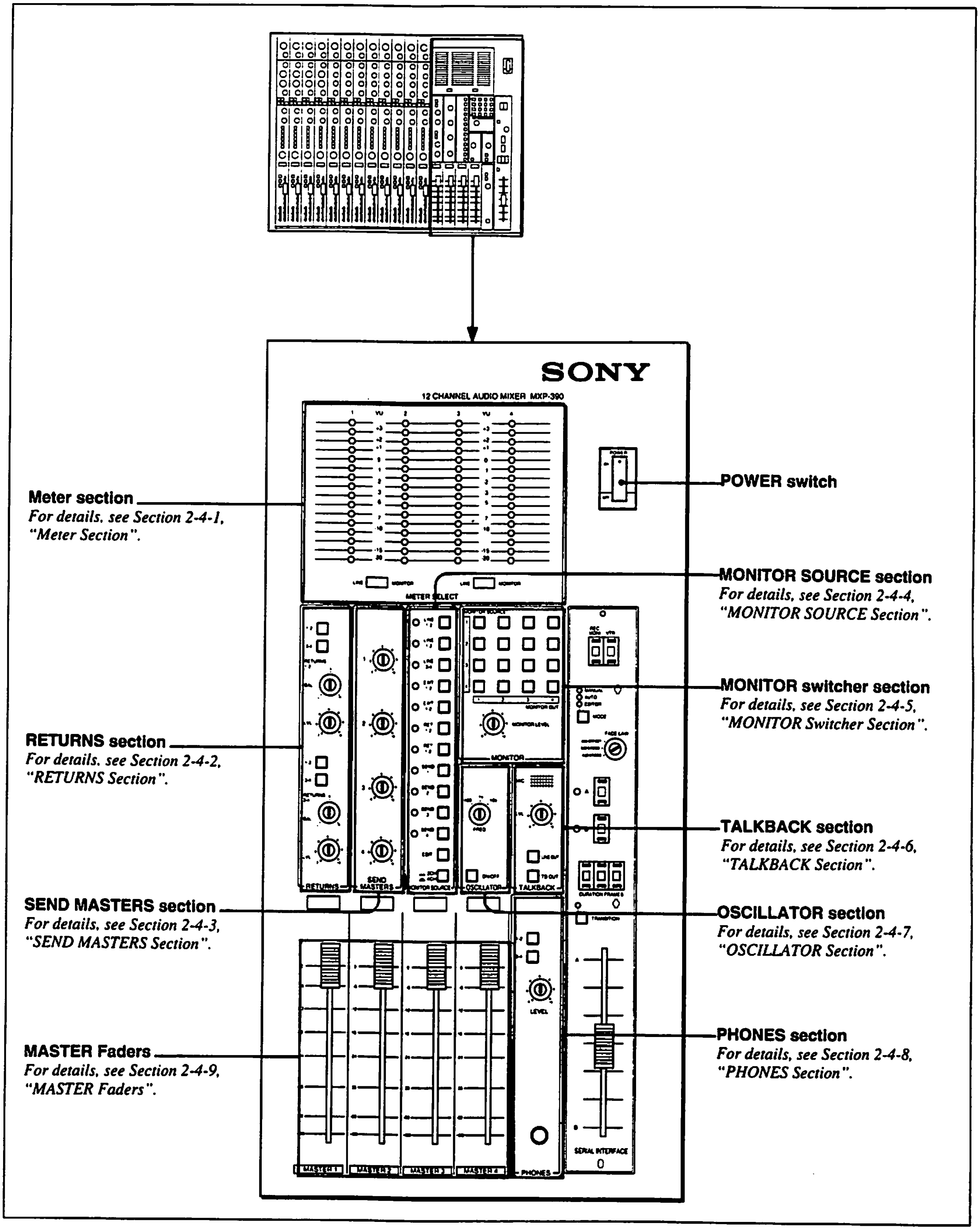
Control panel

The MXP-390's control panel consists of four sections as shown below.



Control panel

2-1 Construction of Control Panel and Connector Panel (Continued)



Output and monitor signal section

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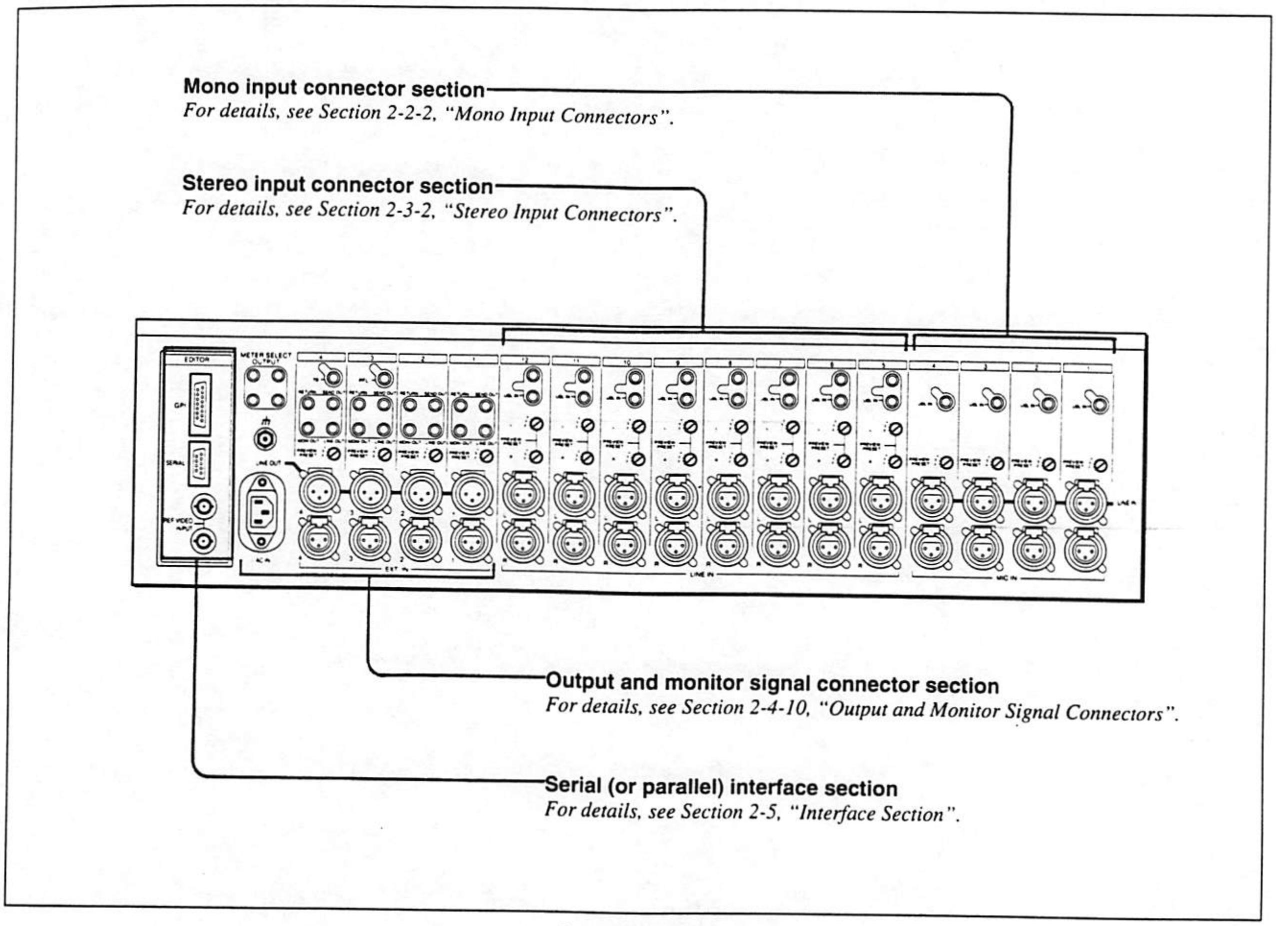
Connector panel

The MXP-390's connector panel consists of four sections as shown below.



Connector panel

Time Code

Time code's sole purpose is to distinguish when one image occurred in time relative to another. Time code tells you *when* an image happened.

If your life was time coded then you would simply know that on April 2nd, 1982, at 10:31PM and 33 seconds you were eating a forkful of spaghetti. This is to distinguish from April 2nd, 1982, at 10:41PM and 15 seconds when you were also eating a forkful of spaghetti but now the plate it empty. The image looked the same at both instances... almost. When each event has its own unique code representing when it occurred, any event can be recalled precisely.

An NTSC video signal is measured in time many ways. It's primary time-base is 30 frames every second. This is like film having a time-base of 24 frames every second. However, video also has 525 lines every frame, and 212.5 lines per field, or 2 fields per frame. All of these time-bases are defined by the electrical video standard (like NTSC, PAL, SECAM).

Time code is field and frame based. Time code is an electrical code that is stored along with each frame or field of video to indicate when it happened. The code is broken down into hours, minutes, seconds and frames.

Just like there is more than one way to encode light information into a single video signal (NTSC, PAL, SECAM, etc.), there is more than one way to represent time code information. There is more than one time code standard.

Fortunately, the Society of Motion Picture and Television Engineers (SMPTE) try to make things more uniform by agreeing upon standards which everyone may use. The SMPTE time code standard is the one that is most widely used by video tape recording systems. However, there are several others. Sony has a time code format strictly used with Hi-8 called RC time code and it is not compatible with SMPTE. To be safe, use SMPTE time code whenever possible. Even Hi-8 video tapes can record SMPTE time code instead of RC time code.

What is time code?
Time code is an electrical signal. Time code is represented by binary (or digital) information. What the electrical, digital time code represents is:

(HH:MM:SS:FF) - (hours:minutes:seconds:frames)

Time code is only useful when it is recorded onto a video tape so that specific frames can be relocated over and over again.

Where is time code?
Time code is written in one of several places on a video tape. Video tape has several types of tracks: audio, video, and control track.

Placing time code is simply a matter of choosing what type of track you would like to record it on. Recording time code onto an audio track means simply taking the digital 0's and 1's and turning it into an electrical signal that can then be recorded by the audio heads of the tape recorder. Recording time code onto the video track means turning the 0's and 1's to electrical information which can be written by the video record head.

Naturally, if there is time code written into the video signal, playing back the video signal will show tiny bright dots (electrical information, high voltage, representing bright light) and dark dots (electrical information, low voltages, representing lack of light). A bright dot is a 1. A dark dot is a zero. Because it is contained in the video signal, it gets transduced back to light along with the rest of the picture information.

Time code stored in the picture information would appear obtrusive if there were bright and dark dots across your picture. Fortunately, there are portions of the video signal that aren't seen but are necessary for the monitor to playback the signal correctly. One of these areas of the video signal is the *vertical interval.* The vertical interval is simply the interval in which the electron gun shoots its way up from the lower right of the screen to the upper left. The amount of time this takes is not something that is *seen* on the screen, but it must happen if the electron gun is to be in the right place to start drawing the next frame. The vertical interval is an ideal place to store any electrical codes that you don't want to be seen by the viewer. By adjusting the vertical scanning of the monitor, you can see the vertical interval. It is a dark bar where there is no picture information. If there is time code in the vertical interval, it will appear as bright and dark dots, flashing on and off as the time code changes.

Time code stored in the vertical interval is called Vertical Interval Time Code (VITC).

If there is time code contained in the audio signal, you will hear the 1's as loud and 0's as silence when the audio track is played back.

Audio tracks are written on magnetic tape longitudinally - directly across the length of the tape. This is means they are called longitudinal tracks. Recording time code on an audio track means recording it longitudinally. Thus, audio time code is given the name Longitudinal Time Code (LTC) because of the way it is recorded. Connecting an audio track or time code track containing time code information to an amplifier will produce the sound of high's (1's) and low's (0's) rapidly fluctuating. This is synonymous with the bright and dark spots representing 1's and 0's in the VITC signal, except you hear the signal instead of see it.

An unforeseen problem with the changing times... harmony
When two vocalist are singing a duet, it is generally appealing if their voices are harmonically related. If they are both singing the same note, it might sound like one person singing. If they are singing notes that are harmonically related, it produces a chord that is pleasant to hear, because a third note is produced from the difference of the two notes they are singing. The third note is also harmonically related to the two notes the singers are singing. However, if the two notes they are singing are not at all related, the third note produced is also not related, and can sound obtrusive.

When the NTSC signal was first standardized, it was an electrical signal that represented only black and white information. It represented brightness values in a scene over time, but not color information. The NTSC standard stated that there would be 30 frames per second, of 60 fields per second - 2 fields per frame. This was based partially on the fact that the electrical alternating current (AC) standard in America was 60 cycles per second (60 Hz). Having the electrical video signal match the electrical AC coming into the walls of every American home makes sense. The two frequencies match and so there is no worry of interference between them. There will be no obtrusive frequencies produced if the video signal and the electrical AC signal interfere.

However, when color TV was invented, a way was needed to add color information to the luminance channel. Luminance information was carried on one carrier. Color information was sent along with it on a *color subcarrier*. The color subcarrier must be harmonically related to the luminance channel, and the frame/field rate, thus related to the electrical AC signal coming into the outlets of a house. A frequency was selected that would satisfy all of those requirements: 3.58 MHz. So color information is *carried* on a color subcarrier vibrating 3.58 million times a second. However, this caused slight distortions of the field/frame rate! To make everything harmonically related, the frame rate was slowed from 30 Hz to 29.97 Hz, and the field rate became 59.94 Hz. This slight slowing of the frame rate made the signal harmonically related to the 3.58 MHz subcarrier.
The result was a luminance channel and a color channel carried on two different carriers that barely interfered with one another... and when they did it was not as obtrusive as if the two carrier frequencies were harmonically unrelated.

This was not a problem until the invention of time code. Time code is based on a system in which there are 30 frames every second, just like black and white TV. However, the color NTSC video signal is not really 30 frames per second but 29.97 frames per second. After 100 seconds (30 x 100) 3000 frames have passed in the time code counter. After 100 seconds (29.97 x 100) 2997 frames have passed in the NTSC color video signal. This means that at frame 2997, the time code claims 3000 frames have passed! An error! Trusting the time code at this point is giving you an inaccurate impression of what frame you are actually at. On a master tape where the actual length of the program is crucial, like with TV shows or commercials, this is unacceptable.

Drop-frame time code was invented as a means of compensating for this error. Every minute of time code, two frames are dropped from the *time code* counter... except every tenth minute. Drop-frame time code is unnoticeable, then, except at the transition of every minute except every tenth minute. This system keeps the time code accurate over long stretches of time.

Every minute except every tenth minute, 2 frames are dropped from the time code.

01:00:00:29 - 01:00:01:00 - 01:00:01:01... 01:00:29:28 - **01:00:29:29 - 01:01:00:02 -**
01:01:00:03 - 01:01:00:04 - 01:01:00:05... 01:01:29:28 - **01:01:29:29 -**
01:02:00:02...
01:09:00:03 - 01:09:00:04 - 01:01:00:05... 01:09:29:28 - 01:09:29:29 - 01:10:00:00...
01:19:00:03 - 01:19:00:04 - 01:19:00:05... 01:19:29:28 - 01:19:29:29 - 01:20:00:00...
15:00:00:02 - 15:00:00:03 - 15:00:00:04... 15:00:29:28 - **15:00:29:29 -**
15:00:01:02...
15:09:00:02 - 15:09:00:03 - 15:09:00:04... 15:09:29:28 - 15:09:29:29 - 15:10:00:00...

Non-drop frame time code is simple. No frames are dropped. When using color NTSC video signals with non-drop frame time code, an error of 3 frames accumulates for every 100 seconds. Non-drop frame time code is used whenever keeping real time accuracy is not crucial (when not producing a show where the time code *must* represent real time).

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THE HISTORY OF TIME CODE 1

VIDEOTAPE

In its early years, television was LIVE. Programs were broadcast directly from TV studios, and all across the country, people watched the same programs at the same time. But the TV industry wanted to delay programs for broadcast on the West Coast.

In 1956 this problem was solved by the development of the first videotape recording machine (VTR). Programs were produced live in New York, recorded on videotape and fed later to the West Coast. TV had discovered a new flexibility and a growing number of programs were Pre-Recorded.

As more complex productions were planned, the industry developed new methods to log and index the material on the recorded videotape reels, and pioneered techniques to edit the "taped" programs accurately.

FINDING THE MATERIAL

At first, recorded material was located by setting the mechanical footage counter on the videotape machine to a fixed reference point. This method wasn't very accurate, however. There are no sprocket holes on videotape, as there are on film, to keep the frame count consistent. The footage counter can slip, and the videotape itself stretches with use, adding to the possible error.

Material can also be located by counting Control Track pulses. The Control Track is a series of equally spaced,

identical electronic "pulses" that are recorded on the videotape and used to synchronize the VTR machine internally when the tape is played back — a sort of "electronic sprocket hole".

But, like a footage counter, counting the Control Track pulses also depends on a fixed reference point. If the videotape machine loses count due to noise or distortion in the pulse signal, the reference point is lost and the videotape must be taken back to the original start point and re-set.

EDITING VIDEOTAPE

In the 50's, videotape was edited like audio tape, by physically splicing the tape together. Material was located with the mechanical footage counter; the tape was examined under a microscope to find an edit point between frames; the tape was cut with a razor blade and then spliced together.

These edits could be very sloppy, sometimes causing picture break-up or roll-over at the cut, and there was no way to preview an edit decision before cutting the tape.

Nevertheless, videotape splicing was the standard until electronic editing was introduced. Using this new technology, the original source reels of videotape were no longer cut apart. Instead, the source material was electronically re-recorded onto a clean reel of videotape to assemble the final edited program.

The early electronic editing methods were haphazard at best and were nick-named "Edi-Crash" since it was almost impossible to tell exactly where the edit would occur.

In 1963, Ampex Corp. refined electronic editing and introduced Editek, a system that controlled the edit point electronically.

The Videotape Operator would press a button which recorded an audible tone or "cue beep" in an audio channel of the videotape at the desired edit point. The VTR machines were cued back and the edit was

electronically performed by the Recording VTR when it read the cue beep.

But, Editek wasn't frame accurate, and the process was slow, sometimes requiring that a cue beep be erased before another beep was recorded for the next edit.

A problem common to these early electronic editing systems was that they relied on the Control Track pulses mentioned earlier. Any difficulty with the pulses meant problems in the edited materials.

For example, if the playback VTR machine mis-read its pulses during the pre-roll, it might drift out of sync with the record machine and the resulting edit would be incorrect. Even under the best conditions, this type of editing was not frame accurate. The source material could slip, creating a potential error of several frames for each edit.

TIME CODE

In 1967, a new editing method was developed to avoid Control Track editing problems. This method used Time Code to define and identify each video frame with a unique identifying number broken down into **HOURS:MINUTES:SECONDS:FRAMES.**

Time Code made videotape editing more efficient than it had ever been before. In addition to speeding up the editing process, Time Code Editing was frame accurate. It allowed each edit to be previewed and repeated, with all materials remaining in sync.

The editing process was further refined in 1972, with the introduction of computer controlled systems. The most sophisticated computers identified edit points with Time Code numbers, stored lists of edit decisions in memory and performed these edits automatically from the stored Time Code information.

Today's editing technology is based on the use of Time Code.

2

THE BASICS OF TIME CODE

THE TIME CODE SIGNAL

SMPTE Time Code is an electronic, digital signal that is recorded along the length of an audio channel or cue track of a videotape in the same way an audio or sound track is recorded. The Code is electronically synchronized to the beginning of each video frame, making it impossible for the Code to slip.

Each video frame is tagged with a unique identifying number called a Time Code Address. This Address is an 8 digit number representing **HOURS:MINUTES:SECONDS: FRAMES.** The Code also contains other information that will be discussed later.

The total of all Time Code information recorded for each video frame is called an electronic Time Code "word". Each word is divided into 80 equal segments called "bits", numbered "0" to "79". These 80 bits are spaced evenly over the entire video frame, so for every frame of video, there is a corresponding Time Code Address.

Information is written in the Time Code by defining the bits as binary "Ones" or "Zeros". It is this binary arithmetic that a Time Code Reader detects to display the recorded information. 202

Electronically, the bits are created by fluctuations or shifts in the voltage of the Time Code signal. In Figure 1, the signal appears as a toothed line called a Square Wave.

By definition, a new bit, equal to a binary Zero, is create when the signal shifts either high or low, up or down.

Every bit in Figure 1 is a binary Zero. There is only one shift of voltage to create each bit in a single bit period. (⎍ or ⊔)

Figure 2, on the other hand, has a pattern of binary Ones and Zeros in its signal. To create a binary One, there is a second voltage shift halfway through the bit period. (⎍⊔ or ⊔⎍)

This unique method of encoding information is called bi-phase modulation and allows the code to be read forward or reverse, at fast or slow speeds.

Figure 1

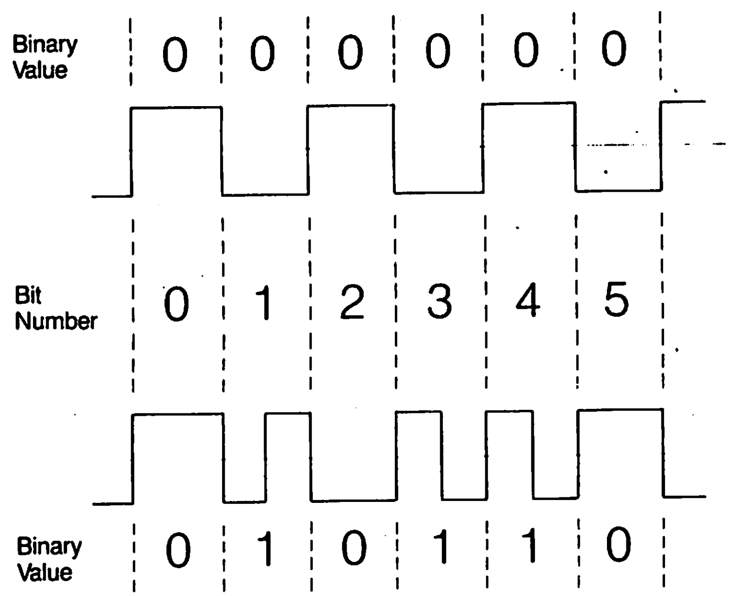

Figure 2

The 80 bit Time Code word is divided mainly into groups of 4 bits to represent different coded information. See Figure 3. Each 4-bit group is coded to form a decimal number (0 to 9), a method of coding called "BCD" or Binary Coded Decimal.

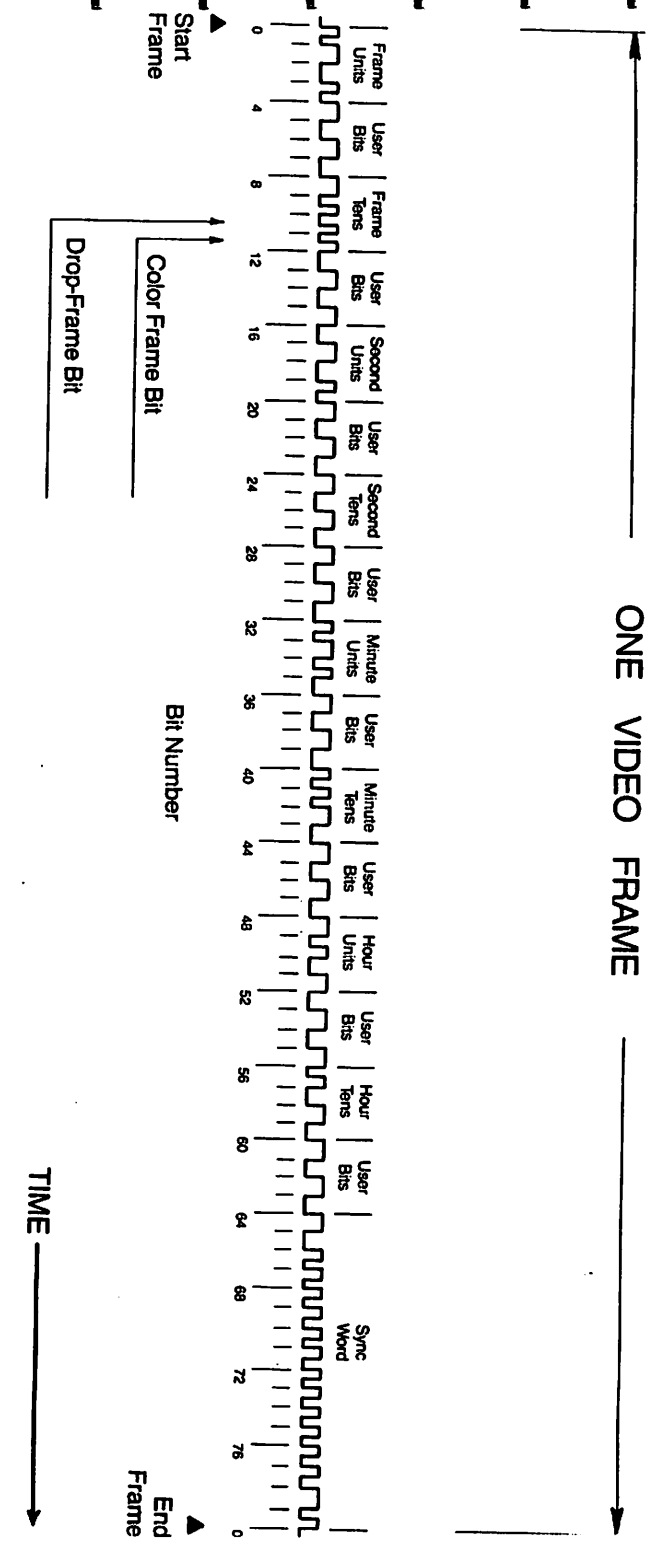

Figure 3
80 BIT TIME CODE WORD
Address Number — 12:35:08:29

203

... a ... Code ... identifies the pattern of binary Ones and Zeros in a 4-bit group, it reads the information as a single decimal number. Eight of these decimal numbers together form the 8 digit Time Code Address, written in an **HOURS:MINUTES:SECONDS:FRAMES** format.

In addition to the Time Address information, there are two other types of information coded into the Time Code word, User Information and Sync Information.

The features of the three types of coded information are discussed in the following pages. For a detailed examination of the Binary encoded data, refer to Chapter 4.

TIME ADDRESS INFORMATION

As described earlier, each video frame is identified by a unique, 8 digit Time Code Address, representing **HOURS:MINUTES:SECONDS:FRAMES.**

The Time Code Address was given an 8 digit "time" format because this format provides a number of advantages in producing and editing video materials.

First, the Address can represent the actual clock time when that particular material was recorded. This is helpful when making a "shot list" or writing a chronological production report.

(It is important to note that Time Code can be used as an accurate clock only if the Code signal is altered. See the discussion of Drop-Frame Time Code on Page 21.)

Second, the duration of a scene or completed program can be calculated with frame accuracy using simple arithmetic.

Third, individual scenes or takes can be identified by programming a specific number in the Time Code Address. For example, when Scene 11 is shot, the Address can be pre-set to 11 hours — 11:00:00:00.

The Time Code Address numbers are recorded in sequence; for each video frame, the Time Code number is advanced by one frame "count".

In the American TV system, there are 30 (or approx. 30) video frames per second (Fr/s) therefore the Time Code Address counts Frames from "00" to "29" and returns to "00" when the Seconds column advances.

In the European system, or a TV system using 25 Fr/s, the Time Code counts frames from "00" to "24" before the Seconds advance.

Time Address information is written into 26 of the 80 bits in the video frame's Code word. Because Time Code runs on a 24 hour clock, there are no hours larger than 23, no minutes or seconds above 59 and no frames above 29. This means the bits assigned to 4-bit Time Address groups are not all needed to encode the Address information. The remaining 6 bits in the Time Address groups are discussed on Page 21.

USER INFORMATION

In addition to the Time Code Address used to index each video frame, there are 32 bits in the Code word set aside for anyone using Time Code to enter their own information.

These "User Bits" were made available for any data the User may require, with no restrictions imposed by the Standards Committee.

User Bits are currently being used as static identification tags for Reel Numbers, Shot or Take I.D., Location, Date or Personnel Data.

But, because Time Code equipment handles User Data with the same techniques as Time Address Data, the User format is limited to 8 digits, and each User digit is limited to the numbers "0" to "9". It can be difficult to encode all the required User information in this limited format.

.. xam , 12 0 28 m , pre Reel . shot
Location #4, Scene 3, shooting on the 28th day of the month. Once set, this particular User data would be encoded into every Time Code word unless manually re-set by the operator.

With a more sophisticated system, each User digit is expanded to a total of 16 possible characters by including both "0" to "9" and the letters "A" to "F". However, the format is still limited to only 8 digits — 2E 36 A4 07.

(The use of four binary numbers in a format to indicate 16 characters is called Hexadecimal Notation.)

Some proposed methods of encoding and decoding User Bit Data will make it possible to go beyond these limitations. For example, User Data may be stretched out over the User Bits in any number of frames until all the required information is encoded.

Larger data capacity will also make it possible to program complete text information, captions, instructional materials, etc., into the Code.

Work is currently underway to propose standard methods of encoding and decoding this expanded data capacity in the User Bits.

SYNC INFORMATION

The third type of information in a video frame's Code word is called "Sync" data. It is an integral part of the Time Code signal and performs two major functions.

The Sync Bits define the end of each frame of Time Code and because Time Code can be read in either direction, the Sync data notes whether the videotape is moving forward or reverse.

The operator has no access to Sync information, which is controlled by the Time Code Generator. Sync data occupies the last 16 bits of the 80 bit Code word.

UNASSIGNED BITS

By the current count, 26 Time Address Bits, 32 User Bits and 16 Sync Bits, there are still 6 bits remaining to complete the 80 bit Time Code word. Four of these 6 bits are unassigned and are defined as permanent binary Zeros.

These Unassigned Bits are available if the need to indicate some standard "mode" of operation occurs in the future.

Since Time Code was first introduced, two modes of operation have been defined and assigned their own bits. A binary One in Bit 10 indicates "Drop-Frame" mode and a binary One in Bit 11 indicates "Color Frame" mode.

DROP-FRAME TIME CODE

As detailed earlier, the Time Code signal is locked to the advancing video frames, that is, Time Code and video frames advance at exactly the same rate.

For Black and White (Monochrome) video signals, this rate is 30 Frames per second. If a Monochrome TV Program is measured by Time Code, everything will agree: the program length, the Time Code display and most importantly, the clock on the wall.

However, Color Video signals do not run at the same speed as Monochrome. The National Television Standards Committee set the Color rate at approx. 29.97 Frames per second (Fr/s).

This means that a Color program, clocked at 30 Fr/s picks up an extra .03 Fr/s every second. Over the course of an hour, the Color program's length expands by 108 video frames, a total of 3.6 seconds.

This accumulated error means that a one hour Color show, measured by Time Code frame count, actually runs 3.6 seconds longer than an hour. There is no agreement between the Time Code display and the clock.

To restore the agreement, the TV industry has developed way to adapt Time Code to make it match the slower running 29.97 Fr/s Color video signal.

The problem is how to lose 108 video frames every hour and still run the Time Code at a consistent speed. The solution is Drop-Frame Time Code.

Instead of the Time Code Address numbers always advancing by one frame, the counting is altered so that whenever the Time Code ends a minute, it drops the first two frames beginning the next minute.

For example, 12:25:59:29 advances to 12:26:00:02. Code numbers 12: 26:00:00 and 12:26:00:01 don't exist

This omission doesn't affect the video pictures. The video frames still progress in a continuous sequence at 29.97 Fr/s. Only the Time Code's number count has been altered; it drops frames in order to run at the same rate as the clock.

However, if 2 frames are dropped every minute for a full hour, 120 frames are lost. Since the goal is to lose only 108 frames an hour, a compromise is necessary — the Time Code is allowed to include the first two frames, but only every tenth minute.

For example, 12:29:59:29 advances to 12:30:00:00 just as it always has.

To summarize, Drop-Frame Time Code eliminates two frames each minute except for minute 00, 10, 20, 30, 40 and 50.

Drop-Frame allows the Time Code to run at almost exactly the same speed as the clock. Any calculations of program length based on Drop-Frame Code will give the correct running time, and both the program length and Time Code display will agree with the clock on the wall

Because the Color TV signal is not exactly 29.97 Fr/s, Time Code running in the Drop-Frame mode still accumulates a slight error, a maximum of 75 milliseconds (75 thousandths of a second) over the course of 24 hours. This error is within the timing requirements of the average production or broadcast facility. For information on Short Term Error in Drop-Frame Time Code refer to Chapter 6.

The old style, continuous Time Code is now called Non-Drop Frame Time Code and Bit 10 of its Code Word remains a binary Zero. Drop-Frame Time Code has an indicator flag built-in by encoding Bit 10 as a binary One.

COLOR FRAMING

Because of the complex way that color information is encoded in a full bandwidth, color video signal, all video frames are not alike. Put simply, video frames have an alternating "characteristic" that can be described as plus or minus, "A" or "B".

When color video material is being edited together, this alternating A-B-A-B sequence of frame characteristics must be maintained.

If an attempt is made to edit frames together out of sequence, that is A-A or B-B, the videotape machine will correct the error by displacing the video picture horizontally. In effect, the Horizontal shift or "H" shift is caused by the system changing the incoming video frame's A or B characteristic to make it fit the A-B-A-B sequence.

Normally, the "H" shift is not visible because it occurs at an edit point between two different shots and the changing picture content obscures the small shift. On the other hand, if the "H" shift is recorded at an edit point in which the picture content does not change, such as a "Match Frame", the shift is visible on the TV picture.

To avoid these visible "H" shifts, it is necessary to identify the characteristics of the video frames involved in each edit. Unfortunately, measuring the A or B of a given frame directly from the color video signal is made very difficult by the distortions present in VTR recording.

For this reason, a Time Code "Color Frame" standard was developed to allow a Videotape Editor to make the A-B frame identification from an easier source — using the video frame's Time Code Address number instead of the video signal.

A Time Code Address ending in an even number is defined as a video frame having an "A" characteristic. An Address ending in an odd number is defined as a "B" video frame.

Once the A or B characteristic of recorded video material is identified, the videotape can be synchronized to a stable reference signal such as the house Sync Generator. This allows the Computer Editing System to insure that the A and B values of incoming video material will match the A-B-A-B sequence already recorded on the edited Master tape.

Whenever a Time Code Signal identifies this A-B color frame characteristic, a flag is raised by encoding Bit 11 of the Time Code Word as a binary One.

3 TIME CODE APPLICATIONS

PRODUCTION AND BROADCAST OPERATIONS

Time Code equipment is light and portable, providing an accurate reference in the studio and on location.

Code can be generated at Time-of-Day (TOD) as a substitute for clock time, or any specific time can be set on the Time Code Generator as required for sequential, multiple-reel productions.

Within a given reel, Time Code simplifies logging and indexing since the Time Code number is always the same for its corresponding video and audio information. Shot and Take notes can be extremely accurate. Lapsed time and time remaining on the reel are easier to determine, and final lengths are available with a frame accurate count, well beyond the limits of a stopwatch.

Time Code is especially important to TV news operations where recorded material must be handled rapidly to meet daily deadlines.

Specific shots and takes are located quickly in the mass of raw footage from the field using a list of Time Code Address numbers; frame accurate "In" and "Out" points are chosen; the news segment is edited and its final duration is logged. The segment's "In" Address then becomes cueing information for the Videotape operator.

Time Code also serves news and other production operations as a library index for storing historical footage.

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VIDEO...PE EDITING

Once program material is recorded, Time Code is the too used "Off-Line" to prepare for Computer Editing and to actually perform the "On-Line" computer program assembly.

Time Code exists in the original material as an electronic signal on the audio or cue track, and the first step in Off-Line editing is preparing work materials that include the Time Code numbers.

The original material is dubbed, usually to 3/4" videocassettes, recording the corresponding Time Code on the outside audio track or Audio 1, and the program sound on the inside track, Audio 2.

At the same time, a Character Inserter, usually built into the Time Code Reader, records a visible display of the Time Code over the picture on the videocassette, as shown on this Handbook's front cover. The size and placement of this "burned-in" Time Code display is selectable and the Time Code numbers are usually surrounded by a dark "box" to help them stand out fro the video picture.

The resulting videocassette now contains every frame of video/audio, identified by its permanent Time Code number both visually and in an audio track. It is a work picture in which the video frames and their Time Code Address numbers are identical to the original material.

Post Production editing is performed using the work videocassettes with the burned-in Time Code display. This is the most economical method of video editing, since at current prices, an hour Off-Line with a 3/4" videocassette editing system costs less than 1/3 the price of an hour On-Line in a professional Computer Editing Suite for 1" or 2" videotape.

The actual edits performed by most 3/4" vi assette editing systems are not frame accurate, but that is not a factor in the final assembly of the program On-Line.

Frame accuracy comes from the burned-in Time Code display on the videocassettes. This display can be read easily at slow shuttle speeds and in still-frame, making the Time Code Address numbers a quick reference.

The production team takes careful notes from the display to develop a list of Address numbers that accurately reflect each edit.

This list of numbers is called the Edit Decision List or Edit List. If the videocassette editing system is not frame accurate in a particular edit, the error can be corrected on the Edit List.

When the Edit List is programmed into the On-Line Computer Editing System, all the final program edits will be frame accurate, exactly as they are written in Time Code Address numbers.

COMPUTER TIME CODE EDITING

Before editing can begin, a reel of videotape must be prepared to record the final edited program.

Time Code and other reference material known as "Basic" or "Edit Black" is recorded along the entire length of the Master videotape with no gaps or breaks in the signal. The "Basic" Time Code numbers will become the final identifiers for every frame of video recorded into the completed program.

Next, the edit list is typed into the Computer Editing System's memory on a computer keyboard and any obvious problems in the list are cleaned up.

Given the proper instructions from the Videotape Editor, the computer-controlled 1" or 2" editing system refers to Time Code to perform each operation. The computer synchronizes all the equipment being used, controls

multiple previews for edit rehearsal, pre-rolls all video and audio machines, rolls the machines in sync, performs the edit with frame accuracy and re-plays the edit for final approval.

Most computer editing systems operate in two modes. One provides manual control for each edit. The second mode called Automatic Assembly, will continue editing working through the Time Code Edit List until told to stop.

The newest editing systems use Time Code Address numbers to control switchers, character generators and other equipment.

Program Materials can be edited; recording both video and audio, or the video can be completed first and the audio added later in a Time Code synchronized pass for sound mixing.

AUDIO RECORDING

Time Code is a recent addition to audio production facilities where it is used as an accurate sync reference for audio recorders and an index for locating sound records on audio tape.

Many audio studios synchronize multiple machines with Time Code. It is also used as a reference in multi-track recording and to prepare cue sheets for later mix-down of the final product.

Audio studios handling film, TV, and soundtrack work use Time Code to synchronize picture and sound for lip-syncing, overdubbing narration, music or sound effects.

In audio editing situations, Time Code provides the ability to preview audio edit decisions, and make accurate on-the-word edits.

Applications for Time Code in the film industry are still largely experimental, although a number of possible uses are being investigated.

Time Code Generators built into film cameras and sound recorders are one option, allowing the Time Code to be laid on the film in the optical or mag stripe, and on a sync or audio channel as the film sound is recorded.

The camera and sound recorder must be synchronized at the beginning of each day's shooting and run on Time-of-Day Time Code, allowing anything shot or recorded all day to be synced to their common TOD Time Code Address.

Another proposal is to use the Time Code as a sort of electronic latent edge numbering. This version of Time Code doesn't include frames, but both picture and sound will still have the Hour:Minute:Second Code in common for syncing.

A third option is to edge number the film as it always has been done, but instead of editing the film itself, the raw footage is transferred to videotape.

At that point, the latent edge numbers from the film are encoded in the Time Code's User Bits, establishing a permanent relationship between film and video frame numbers.

Once the material is video edited and the final cut approved, the video Time Code numbers are converted to film edge numbers and this list is sent to the film negative cutter to prepare the original film materials for printing.

The major problem with this option is the difference between film running at 24 Fr/s and video running at 30 Fr/s. To make each second of film equal each second of video, the film frames are scanned at an uneven rate during their transfer to videotape. This causes problems for the negative cutter, who works from an edge number list that is calculated from the Time Code.

Time Code does offer a number of advantages for film, many resulting in savings of time and money. Since Time Code is a frame accurate reference, clapstick slates are eliminated, saving on raw stock; picture and sound for dailies (or rushes) can be synced quickly and easily; preferred takes are easier to find and log; and any number of cameras and sound recorders will always be in perfect sync.

The most important applications for Time Code may be new techniques for editing film.

The process described earlier — editing on videocassette and conforming a final film print for release — represents one possible combination of film and video technologies.

There are also editing systems that use traditional film work prints and film transports (like Steenbeck or Moviola), but play the images back through a sophisticated control panel and record the edits on videocassette.

But the most common adaption, especially in films produced specifically for TV, is to transfer the film footage to videotape directly from the original negative. The latest Telecine Cameras and "flying-spot scanners" can accept negative image and feed a positive image that has been scene by scene color balanced. Most of these devices use Time Code as a reference.

The resulting material looks like original film, but is recorded on videotape and is Time Code identified frame for frame. Work cassettes with burned-in Time Code are usually made during the film transfer, so all picture and sound work can be performed on videotape.

Regardless of how the final picture will be seen, on film or video, Time Code is the most accurate method of producing film sound, from building dialog, music and effects tracks to syncing all the sound materials on a multi-track machine for the final mix.

This new ability to combine film and video materials means that mixed-media sync can be accomplished easier than ever before, through the use of Time Code.

4 SETTING UP FOR TIME CODE OPERATION

TIME CODE GENERATORS

A Time Code Generator is the device that creates the digital Time Code signal. The most common Generators include a front panel display for the Time Code data and a method of pre-setting and starting the Code, such as thumbwheel switches.

Normal procedure is to set the Generator to Time-of-Day Code, although on most Generators, any time can be selected from 00:00:00:00 to 23:59:59:29.

Generators may provide a remote control switch to run and stop the Time Code. This may be necessary if the Generator is to be mounted in a rack away from the studio or videotape room.

One very important feature of a Time Code Generator is the "Jam Sync" function, which allows the Generator to pre-set or "Jam" its Time Code output to an external Time Code signal. The major use of Jam Sync is "restriping", a process that replaces deteriorated or discontinuous sections of Time Code on a videotape with new Code that matches the original Address numbers and frame count.

SYNCHRONIZING THE TIME CODE

To insure that the Time Code is locked to each video frame, the Time Code Generator MUST be synchronized to the video source being recorded.

In a house system in which all equipment is "Gen-locked" to a common video Sync Generator, all that is necessary is a separate sync feed to the Time Code Generator.

If no "house sync" is available, Time Code Generators are designed to accept any continuous composite video or sync signal which is fed to the Time Code Generator from the video source being recorded.

Failure to supply the Generator with a proper reference signal will result in recording non-synchronous Time Code, which is useless for Computer Editing and most other Time Code applications.

Figure 4 shows a basic system diagram including the Time Code Generator.

DROP-FRAME AND COLOR FRAME MODES

There are two modes available for Time Code operation.

Drop-Frame Mode
Drop-Frame Time Code is the accepted standard in the majority of modern production facilities. Selection of Drop-Frame Mode is accomplished by an internal or external switch on the Time Code Generator.

Any facility using Time Code as clock time, or editing and calculating the length of TV programs should use Drop-Frame.

Audio Production facilities that at some point may synchronize sound to video should also select Drop-Frame.

Color Frame Mode
Facilities editing full bandwidth, 1" or 2" videotape should use Time Code generated to the Color Frame timing standard.

Generators capable of creating Color Framed Time Code will either require an additional "Color Frame" input fed from the house Sync Generator (normally a 15 Hz signal), or a specially adjusted "RS-170A" video feed.

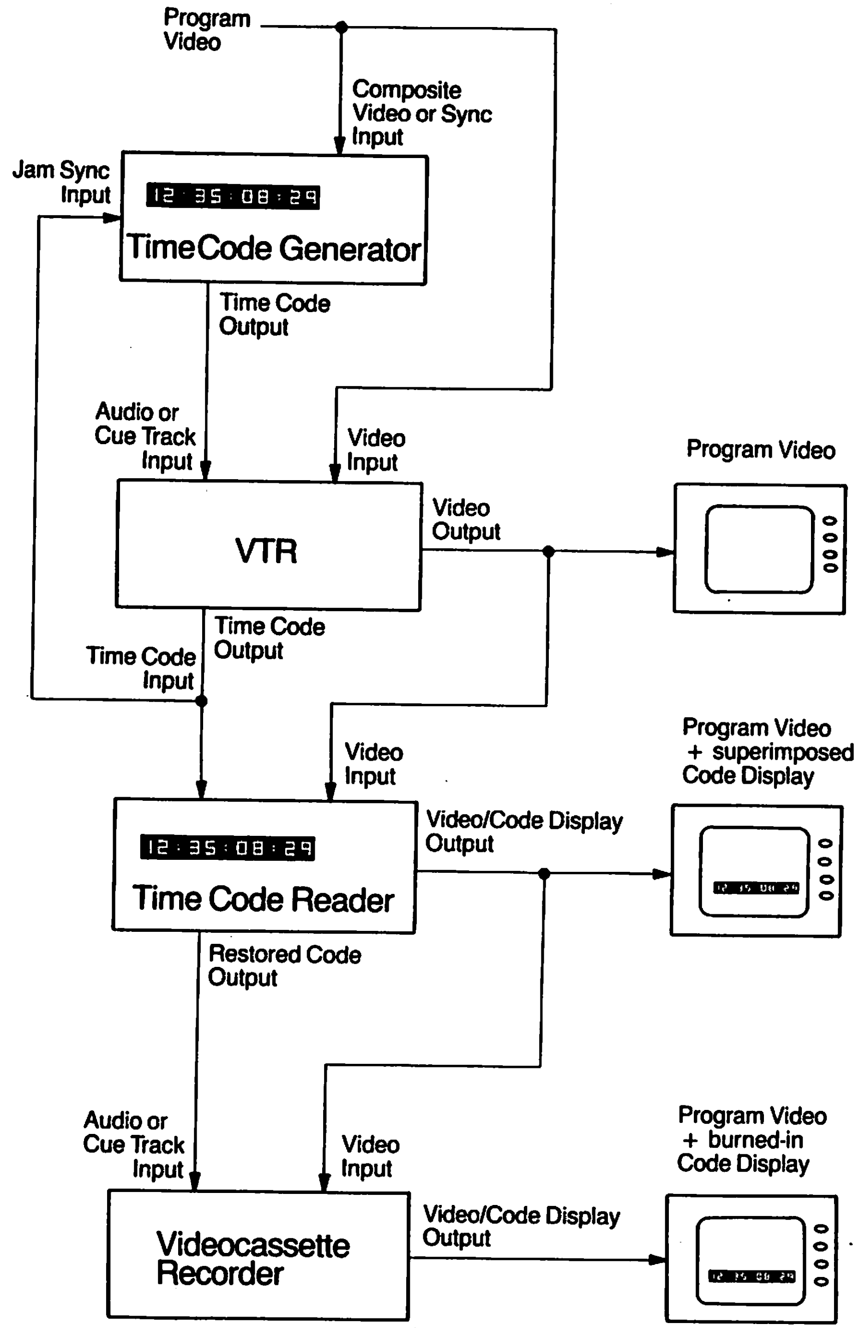

Figure 4 — SYSTEM DIAGRAM

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There are a variety of frame rates selectable in most Time Code Generators.

Color videotape and TV operations require the NTSC rate of 29.97 frames per second.

Monochrome (B + W) TV and some audio studios use 30 frames per second, although it is suggested that audio studios use the 29.97 color standard, for later synchronization with video.

Time Code used in connection with Film will usually run at 24 frames per second, with the capacity to vary the Code timing to match film camera speed.

European Broadcast standards require 25 frames per second to be compatible with the European TV signal.

DISTRIBUTION

Time Code may be treated like any other audio signal. It can be routed through audio Distribution Amplifiers and two-conductor shielded audio cables, and patched through audio Switching Systems.

A potentially serious problem is Cross-Talk caused by the Time Code signal interfering with adjacent, low level audio signals. As with any high level signal, Time Code cables should not be placed too close to other audio routing lines.

Since Time Code is a symmetrical signal, it is immune to normal cable polarity problems, and is unaffected by inverted or "flipped" lines.

The digital Time Code signal, like an audio feed, is recorded on an audio or cue track of the videotape.

For 1" helical tape, record the Time Code in the 3rd audio or cue track.

For 2" quad tape, Time Code is recorded in the cue track (also called the Time Code track or auxiliary track).

For 3/4" helical videocassette, use the outside track or Audio 1. Some cassette machines feature a special Address or Time Code track with its own input. This frees Audio 1 for production work.

Most videotape machines have automatic gain or level controls for the audio inputs. These automatic controls should NOT be used for recording Time Code. To avoid distorting the Code signal, the audio gain should be set manually.

There are no current industry standards for Time Code recording levels, but field experience shows that the following levels will give the best results in most cases.

For 1", record the Time Code with a level between −5 and −10 VU.

2" quad requires the highest signal possible without causing problems such as Cross-Talk: +3 to +5 VU.

3/4" cassette machines always require a few tests to determine the best level, but it will generally fall between −5 and 0 VU.

Time Code should be recorded on Audio Tape on the furthest outside track available, saving the more responsive inside tracks for sound work. It is also a good idea to leave a blank track between the Time Code and the recorded sound to avoid Cross-Talk in multi-track audio production.

There are two basic types of Time Code Readers: Playspeed-only and Variable Speed.

Variable speed readers are preferred since, especially in editing situations, it is necessary to read the Time Code across a wide range of tape speeds. Most modern readers can decode information from 1/50th to 100x playspeed. However, having a variable speed reader is no guarantee that the Code can be read at <u>all</u> speeds.

Time Code is recorded along the length of the videotape (in the "longitudinal mode"), therefore the tape must be moving over the playback heads at some minimum speed before there is sufficient level for the Time Code Reader to detect an error-free signal.

This is a problem at slower shuttle speeds and it is impossible to read SMPTE Time Code when viewing a still-frame. On a 1" helical VTR, Time Code is usually read accurately as slow as 1/20th playspeed. Most 3/4" videocassette machines only read accurately to 1/10th playspeed.

When using a Time Code Reader that can decode information at extremely high speeds, the major limitation is the bandwidth of the VTR's audio channel.

Shuttling at many times playspeed, the Time Code signal can go beyond the audio channel's ability to handle the information. The only solution to this problem is to modify the VTR audio (or cue track) playback amplifier to give it sufficient bandwidth to handle a wider speed range. For more information, see Chapter 7.

Another problem at extremely fast shuttle speeds is the "Air Layer Effect" in which the tape is moving so fast, a thin layer of air lifts the tape off the heads, causing the Time Code Reader to receive false information.

lev[illegible] whic[illegible] sig[illegible] as o[illegible] lly r[illegible] [illegible]
Generally speaking, each videotape machin[illegible]s its own "personality" in the manner in which it records and plays the Time Code signal.

A good rule of thumb is to experiment with each machine to determine the minimum record level necessary to assure accurate playback. Playback levels that are too high can cause Cross-Talk and levels that are too low may be impossible for the Time Code Reader to detect, especially when exchanging tapes between different VTR machines or different Readers.

Vertical Interval Time Code is being developed in an attempt to solve some of the above problems, and is discussed in Chapters 5, 9 and 10.

DISPLAYING TIME CODE

The most common display of Time Code information is on the front panel of the Time Code Reader, usually in the form of LEDs (Light Emitting Diodes) that show the Code as it advances frame by frame.

In most cases, however, it is necessary to see the Time Code numbers and the video pictures together. For this reason, most Time Code Readers have built-in Character Generators and Video Inserters, enabling the operator to superimpose the Time Code display over the video picture.

This can be done live without affecting the picture recorded on the original videotape, or it can be used to produce work copy videocassettes which have Time Code permanently burned-in to the picture area.

Both the size of the visible Time Code display and its placement in the TV picture are selectable by controls on the Reader. Visible Time Code is usually surrounded by a dark "box" to make it stand out from the video picture.

displaying the burned-in Time Code in the video picture. For example, an edited 3/4" cassette may be screened for content approval, and visible Time Code would obscure some portion of the picture area.

In this case, some Time Code Readers can position the Code display in the Vertical Interval, outside the TV picture's cut-off area. The edited video picture will be free of visible Time Code, and the burned-in display can be seen only by using a special Pulse Cross Monitor.

If Time Code is to be used for Computer Editing, it is necessary to supply the Code information to the computer. Most Time Code Readers are equipped with outputs to convert Time Code into BCD parallel data that the computer can read.

The types of display activities described above usually require that each VTR machine have its own Time Code Reader. Depending on the production load, an entire facility may or may not require more than one Time Code Generator.

A BRIEF CHECKLIST

- When adding Time Code to an existing system, it is important to check portions of the equipment (such as Time Code audio tracks) that may not have been used before.

  Also, since the digital interface between Time Code equipment and editing systems is not standardized, there may be interconnection problems. Be certain your system is compatible with the Time Code equipment.

- A Video reference or Sync signal <u>MUST</u> be patched into the Time Code Generator, otherwise the recorded signal will be non-synchronous and useless for Computer Editing.

... the recording and playback of Time Code. Dirt, ... distortion and poor levels in the Time Code signal. Carefully clean the record, play and erase heads and the entire tape path prior to each use.

- Treat the Time Code as any other signal and make test recordings, especially to check for proper levels. As discussed earlier, every system will handle Time Code differently.

- Choose either Drop-Frame or Non-Drop Frame Code and stick with it. Attempts to mix these modes can cause difficulties in production and editing.

- In order for a computer to locate material on the videotape, the Time Code should be in ascending order on all reels and the same Code Address number should only appear once on a reel.

- There should be a minimum of 10 seconds of clean Time Code prior to any material included in an edited program. Otherwise, the computer will have insufficient room to pre-roll or back cue the tape to perform an edit.

- Material recorded near midnight may run into a problem. At midnight, the Time Code's clock advances not to 24 hours but to Zero hours. Recording this transition into a reel of videotape intended for editing will confuse the computer since the lower Code numbers will appear at the end of the tape.

  To avoid this, either start recording sufficiently in advance of Midnight to finish before the "Zero Hour" or wait until after Midnight to begin. A third option is to use the Time Code Generator's capacity to pre-set a Code Address independent of the clock.

Additional problems and solutions can be found in Chapter 8.

VERTICAL INTERVAL TIME CODE 5

PROBLEMS WITH SMPTE TIME CODE

The most obvious problem with SMPTE Time Code has to do with reading the Code signal at extremely slow speeds.

There are high quality, variable speed Time Code Readers that can decode from slower than 1/50th to 100x normal playspeed, but because of limitations in the tape system, even they are ineffective at slower shuttle speeds and still-frame.

In addition, the Time Code signal takes up valuable space on the videotape, usually in an audio channel that could be put to better use for sound production work.

For these reasons, other methods of frame identification are being developed.

VERTICAL INTERVAL TIME CODE — VITC

The most promising alternative to SMPTE Time Code was introduced in 1978 and proposed as a standard in a slightly modified form in 1980.

Vertical Interval Time Code contains the same Address and User Information as SMPTE Code, but it is a different type of signal. VITC Code is recorded within the video signal itself, outside the visible picture area in a location called the Vertical Blanking Interval.

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First, on 1" or 3/4" helical videotape, VITC can be read at the slowest frame-by-frame jogging speeds including still-frame. Because VITC is a part of the video information and is read by the rotating video heads, any time a video picture is visible, the Time Code is detectable.

This means VITC can replace burned-in SMPTE Code on work videocassettes. A Time Code Reader can superimpose the frame accurate VITC display when needed to develop the Edit List, but it will no longer be necessary to burn-in a permanent visible SMPTE Code display.

Second, VITC identifies video fields.
As discussed earlier, a SMPTE Code word is spaced evenly over an entire frame of video material. But, each video frame is really constructed of two individual fields, produced as the picture tube scans the odd lines (Field 1) and then the even lines (Field 2) in a video picture.

Since VITC is recorded in the video signal itself, this Time Code indexes both fields of a frame with an Address number that includes field identification.

Field identified Code means a sophisticated computer editing system using VITC can edit with field accuracy. In effect, VITC expands the number of edit points from 30 per second to 60 per second, since there are 60 fields that make up every 30 video frames.

When it is the only Time Code in use, VITC frees an audio channel for production, and completely eliminates the potential for Cross-Talk from the SMPTE Time Code signal.

VITC does have minor disadvantages. For example, some 3/4" videocassette machines not capable of reproducing video at full shuttle speeds will still rely on SMPTE Time Code. 2" quad machines will also continue to use SMPTE Code since quads cannot read still-frame or retain an image at speeds other than normal playback.

Although it does not take up space on the ta ke SMPTE Code, VITC does occupy scan lines in the Vertical Interval that could be put to other uses.

Also, since VITC is a part of the video signal, it is difficult to add after the original material is recorded. In most cases, the video material must be dubbed down a generation to another tape while the VITC Code is added.

One solution to the question of which Time Code to use is: Both. Systems are being developed that could use whichever type of Time Code is detectable in each instance. When SMPTE Code and Vertical Interval Time Code are recorded on the original materials, these systems would combine the strengths of both.

For a more detailed look at VITC, see Chapters 9 and 10. The complete text of the SMPTE Proposed VITC Standard, is available in Appendix B.

ning le va s not same he R r

expected, it will display the expected value rather than the actual decoded value.

This process continues for a preset number of frames as an "error limit", after which the Reader will output the decoded value even if it doesn't match the expected value.

For example, a 2 frame error limit means a disturbance of the Code signal up to 2 frames is not visible in the Time Code display. Beyond 2 frames, the error becomes visible.

The problem with Error Bypass is that valid changes in Code values — for example, a break in the middle of a reel of tape during which the Code changed from 1 to 2 hours — will de delayed during decoding for the duration of the error limit.

Plus-One-Frame

Plus-One-Frame is an important feature on a Time Code Reader since it assures that the Code Address being displayed corresponds to the current video frame.

Not all Readers feature Plus-One-Frame, but it is required for accurate Code display, because of the structure of the Code word.

The Reader decodes information continuously, but it will not update the Code display until the end of the 16 bit sync data at the end of the Code word.

That means all the video information for a frame is already transmitted before the Reader can display the Address number it is reading.

Therefore, there is natural delay between reading the Code word and displaying the Address data.

To compensate for the delay, the Reader must automatically add one frame count to the Address it is reading in order for the Code display to match the video frame being scanned.

Difficu s Re g Co

Code Reading difficulties are mainly a function tape speed, once the Code input is above the minimum required level for the Reader.

Most variable speed Readers are designed to decode bi-phase signals from 1/50th to 100x playspeed. Their ability to read from tape is reduced by physical limitations in the record/playback process.

Still-Frame and Slow Tape Speeds

SMPTE Code is a longitudinal signal requiring a minimum tape speed over the playback head to detect the Code.

SMPTE Code cannot be read at Zero tape speed (still-frame) as encountered on 1" and 3/4" helical VTRs.

At very slow speeds, typically less than 1/10th playspeed on 3/4" VCRs and 1/20th playspeed on 1" VTRs, the reproduced waveform exhibits extreme amplitude distortions that can cause extraneous transitions. These "false" transitions are detected as valid by the Reader's decoding circuitry, and the Reader will output false data.

The solution to these problems is an alternate type of Time Code. See chapters 9 and 10 for information on VITC — Vertical Interval Time Code.

High Shuttle Speeds

Limitations with high tape speeds relate primarily to the high frequency characteristics of the system. A normal 2400 bit/sec signal when shuttled at 50X playspeed rises to a bit rate of 120 K bits/sec for binary Zeros, 240 K bit/sec for binary Ones, equal to audio frequencies from 60 kHz to 120 kHz.

These bit rates/frequencies are well above the frequency response of normal audio equipment.

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The redundancy — the placement of VITC Code [illegible] non-adjacent lines — is required by the proposed VITC Code specification. See Appendix B.

The data carried by VITC is much the same as that of SMPTE Code. Drop-Frame and Color Frame modes, Address data, and User Bits are unchanged from the SMPTE Code data. Since this information is transmitted in both video fields, one of the Unassigned Bits in SMPTE Code is redefined as a Field Identification bit to allow for independent addressing of each field in VITC.

Since the VITC data is part of the video signal, there is no ambiguity in image identification. At all times, the frame being seen corresponds to the number just decoded. Therefore, there is no potential for non-synchronous Code in VITC.

The 90 bits of one line of VITC data are transmitted using a bit rate of 1.79 MHz. As this is one-half of the color subcarrier frequency, the VITC signal can be processed by digital Timebase Correctors as well as analog distribution equipment without distortion.

Code Placement
The user of VITC generating equipment must specify which two non-adjacent lines in the vertical interval are to be used by the Code. The first line that may be used is line ten; the last usable line is line 20. The VITC data will be inserted on the same line in both fields.

The full text of the proposed VITC specification is available in Appendix B.

10 VITC CODE IN PRACTICE

VITC equipment looks almost identical to SMPTE Code equipment. All of the usual functions and most of the operational controls are the same; most of the rear panel signals are similar.

VITC equipment includes Code Generators, Readers and Character Inserters, or some combination of these functions. But, there might also be routing switchers, inserter/keyers, and stripper/inserters in some VITC systems.

ROUTING VITC CODE

In a small system, signal routing is straightforward. The Generator inserts Time Code data into two selected horizontal lines of a video signal using an internal keyer. This VITC coded video can then be distributed as needed before being recorded. See Figure 9.

Coming out of a playback VTR the demodulated video signal is fed to a Reader to decode and display the VITC data. If the Reader requires stable video, it must be connected to a Time Base Corrector. The output of the TBC must be adjusted to pass the needed Vertical Interval lines.

If the Reader produces a character output, it can be viewed directly on a monitor, or recorded on videocassette for Off-Line use.

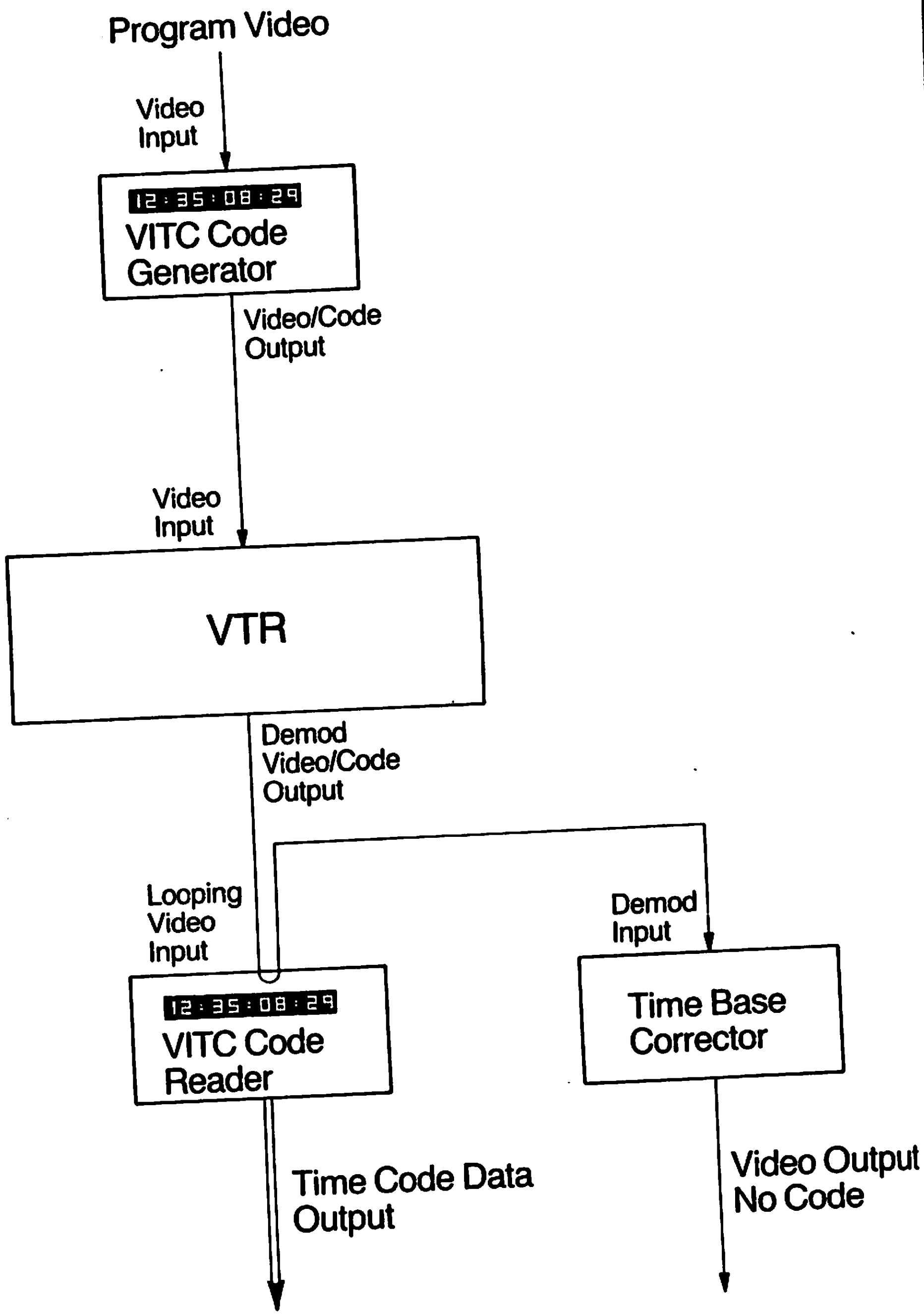

Figure 9 — VITC SYSTEM DIAGRAM

VITC use is more complicated in an editing system. Since you would wipe out any existing VITC on the record VTR each time you make a video edit, the Record machine's Time Code must be reinserted as VITC in its video input. This requires a type of Jam Sync VITC Generator to reproduce the correct data from either the preceding VITC or from a SMPTE Code signal. See Figure 10.

If the VITC Code of the playback VTR is retained in the video, passing through the TBCs and Video Switcher, it can also be recorded on the Record VTR, assuming that non-interfering pairs of Code lines were chosen. This provides a kind of automatic logging of the source and Address of any video edit recorded.

If this logging of video sources is desired but the Switcher and other equipment do not pass Vertical Interval signals, the stripped playback VITC must either be routed around the switcher to a separate re-inserter, or another Generator must be used to reproduce the data on the record video.

DISTRIBUTING VITC

As in current SMPTE Code master Generator systems, it is possible to generate a master VITC signal and then distribute it to many insertion points. The VITC signal is handled in the same fashion as a camera signal, taking into account cable and amplifier delays to produce a properly timed signal at each video inserter.

Considering the relative costs of timed cabling, the inserter or keyer, and the other Generator electronics, dedicated Generator/Inserters are more cost-effective in most cases than a central Generator with distributed Inserters. Dedicated equipment is also more flexible in operation.

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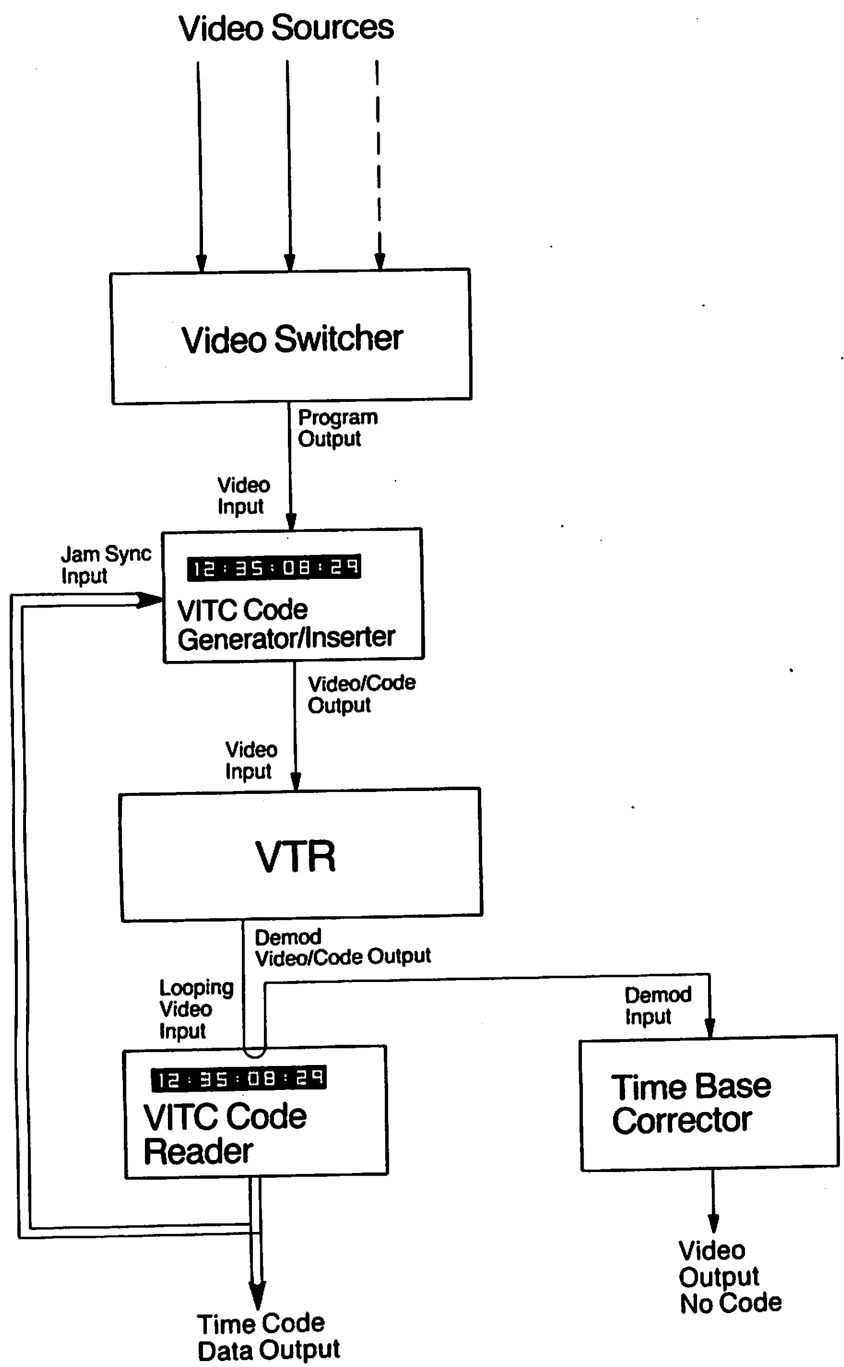

Figure 10 — VITC EDITING SYSTEM

When synchronized timing is necessary, Jam Sync operation to a common SMPTE or VITC feed may be used. Any Generator may be used as the master, or all may be used independently.

RECORDING VITC

As part of the video signal, VITC must be added to a video feed prior to recording. In practice, this means that every Studio output, ENG camera, Network feed, or VTR playback that might be edited using VITC must have the Code added prior to recording because it can't be added later.

For facilities that regularly stripe a tape with SMPTE code during recording or any time thereafter, this is a significant change in procedure.

For some operations, already having material on videotape without VITC, this might mean dubbing pre-recorded footage to gain the advantages of VITC when this material is used in editing. Of course, standard SMPTE Code can still be used if VITC has not been recorded, sacrificing the Reading advantages of VITC at slow speeds.

VITC AND 1" TYPE C VTRs

Although the introduction of VITC was encouraged by the widespread use of Type C VTRs, these machines may provide a difficult environment for VITC use under some conditions.

The Type C specifications allow for optional use of a Sync Head, an auxiliary video record/play head. When used, the Sync Head "fills in" the Vertical Interval dropout caused by the small gap in the tape wrap around the head

um. When the Sync Head is not used, a ten dropout of the video signal is produced, from line 5 to line 15.

Including the problems of tape interchange between VTRs, tracking error, and worst-case "growth" of the dropout with tape motion, the reproduction of lines 3 through 15 cannot be assured without use of a Sync Head.

What this means in practice is that recording VITC on a Type C VTR without a Sync Head limits the choice of VITC insertion to lines 16 through 20.

Also, when recording on a Sync Head equipped VTR, VITC should not be inserted in lines 10 through 15 if Post Production might occur in a "rented" facility not equipped with Sync Head VTRs.

There is one potential benefit resulting from this confusing situation. As noted earlier, VITC usually cannot be restriped into previously recorded material. But, some of the newer Sync Head equipped VTRs have the capability to record a Video insert on the Sync Head only. This allows a video insert edit to be recorded on lines 5 through 16 alone.

As the switching point between Sync Head reproduction and Video head reproduction can "move" in the range of line 12-16, a VITC insert edit on these lines using only the Sync Head might lead to a "hide and seek" situation; the code might appear and disappear as its line is reproduced from one head or the other.

Considering the worst-case situation, only lines ten through 12 seem suitable for adding VITC after video recording.

Before deciding to use this method it is prudent to consider the state of mind required to intentionally press the "Video Insert Edit" button on a reel of irreplaceable original material.

To sum up these suggestions for line usage with Type C Helical VTRs:

LINE #	USAGE
10	save for emergency re-insertion
11	spare
12	save for emergency re-insertion
13-15	not read without Sync Head usable with Sync Head record and playback machines
16	original material VITC
17	edit material VITC
18	original material VITC
19	reserved for VIR video signal
20	edit material VITC

If additional pairs of lines are required, they can be added by working "upward" to lower line numbers, limited by VTR reproduction.

A Note on 3/4" VCRs
These restrictions are much looser in a Cassette operation. All lines in the Vertical Interval are recorded and, except for a "sync fill-in" pulse occasionally found near the vertical sync pulse, all lines are reproduced well.

So, the selection of lines to assign to VITC can be almost arbitrary when recording only on cassettes. Of course, the list above can still be used for compatability with any one-inch material that might be used.

READING VITC CODE

Now that the VITC signals have been recorded on selected lines, how do you choose which Code is to be read? Most Readers have a selection for the lines they will "look at." As long as the lines where the Code was recorded are known, this selection works well.

eve en R ng V rom C o , the ideo might "lift" or misframe by one or two lines if the TBC error window is exceeded by the VTR timebase error. This moves the Code out of the selected lines, producing an intermittent read error.

To solve these problems, various "auto" systems have been developed as alternatives to fixed line selection. Some Readers start looking at the video signal after V Sync and stop when they find data with a good CRC. With several active Codes, this method always finds the Code on the lowest line unless it is obliterated by a dropout. It will then find the next good line, which may not be from the same Code.

When using this type of Reader, always record VITC Code on non-adjacent lines, but arrange the line assignments so that the pairs are not "interleaved."

Another method of automatic line selection reads all Vertical Interval lines, comparing any Codes found to the selected "right" one before display. This method can track a Code that has shifted a line or two, as long as the Code is relatively continuous.

Another line identification problem can arise when the demodulated video output of a Type C machine without Sync Head is fed to a Reader. The Vertical Interval dropout can lead to ambiguous line identification especially at high shuttle speeds. Since the location and size of the line dropout varies, accurate line number determination is difficult.

VITC AND BROADCAST VIDEO

It is unlikely that a VITC signal might be accidentally "aired". The number of TBCs, Processing Amplifiers, and VIR/VITS inserters that the average Broadcast signal passes

thr would seem to insure that the Verti nterval would be scrubbed clean. Occasionally the VIR signal has trouble making it to the transmitter, all effort notwithstanding.

One ready solution to this concern might be to set all TBCs to strip VITC and use Readers that can operate directly from the VTR demodulator outputs.

CAN VITC REPLACE SMPTE CODE?

For those wishing to replace longitudinal Code entirely, an important issue is how fast can VITC be read? The usual answer is, "It depends."

First, the VTR or VCR must be able to produce video at shuttle speeds. For the fastest current machines, this may mean 60-80 times playspeed for a 1" machine and up to 30-40 times playspeed for a cassette unit.

Second, the "doppler shift", or frequency change of the reproduced video signal, must be within the detection range of the VITC Reader. This range is a tradeoff between circuit complexity, stability, and cost, and varies from about 5% to over a 50% acceptance range; about the same as a 2x playspeed to 60x playspeed shift.

(This frequency shift is more severe in one shuttle direction than the other, due to the effect of the head drum rotation.)

Readers using Timebase-Corrected video leave most of the frequency-correction to the TBC, making the TBC correction range the "weak link."

Finally, the Reader has to decode a line of Code without being disturbed by a "hash mark", produced as the video head jumps to the next track. The chance of reading a VITC line without error gets progressively worse above 30-40 times playspeed, so shuttle speed Readers must rely on good error detection and Error Bypass systems to compensate for deteriorating reading conditions.

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MIDI SYNCHRONIZATION AND MIDI TIME CODE

by Paul D. Lehrman

MIDI Time Code has been an official part of the MIDI Specification for over two years now, and during that time it has established itself as an invaluable tool for hundreds of studios. But to the majority of MIDIphiles, MIDI Time Code, or "MTC" — what it is, what it does, and what it's for — is still something of a mystery.

It shouldn't be. Those who use it consider MTC to be the best way to synchronize MIDI systems with the rest of the audio and video world. As MIDI continues to grow and improve, and find more uses in music production, film and video post-production, sound editing, and studio automation, an understanding of MIDI Time Code will be essential to anyone working with MIDI.

MIDI SYNC

Before we get to what MIDI Time Code is, we need to understand what it's for, and why it's necessary. That means looking at its predecessor, MIDI Sync, which has been part of the MIDI spec almost from the beginning.

The ability of MIDI sequencers to follow tape — audio or video — has been an important part of the MIDI revolution. It means that electronic instruments can be used for an almost unlimited number of "virtual tracks" (which do not take up tracks on a multitrack tape), along with "real" instruments (which do), thereby expanding the capabilities

acy epe lity, ithd e ex e an ort of working with tape and live musicians.

The creators of MIDI knew this was important, and developed what is commonly referred to as "MIDI Sync". This actually consists of several different MIDI commands: "Timing Clock", which is a single byte sent 24 times per quarter-note, and therefore changes in speed when the tempo of a sequence changes; "Song Position Pointer", a three-byte message that tells a sequencer where to start, by counting the number of 16th-notes since the beginning of the piece; and "Start", "Stop", and "Continue", single-byte commands which do just what their names say. MIDI Sync is used for more than just locking sequencers to tape: it's also used to get drum machines to follow sequencers, sequencers to follow each other, and even to get sequencers to follow live or recorded performances.

There are two ways to use MIDI Sync (which, for clarity's sake, are often referred to as "Clocks and Pointers") to get a sequencer to follow a tape. The simplest way is with "FSK", an audio signal that moves up and down in pitch very rapidly. To print ("stripe") an FSK track on a tape, you connect a MIDI-to-FSK converter to the output of a sequencer, and tell the sequencer to generate MIDI Timing Clocks as it plays. At each clock (24 times each quarter note), the frequency of the FSK signal shifts. If the sequencer's tempo changes, the frequency shifts happen closer together or further apart. When the sequencer stops, the FSK signal stops shifting.

Now when the tape is played back, the FSK signal triggers an FSK-to-MIDI converter (which is usually the same box, with the "read/write" switch in the "read" position) to produce Timing Clocks and send them to the sequencer. If the tape should speed up or slow down by a small amount, the sequencer will follow right along, and so everything stays in sync.

The disadvantage of FSK is that once it's down on tape, you can't do much about it: you can't change a sequence's starting point, or extend its length, or alter the tempo in any way. If you want to do any of these things, you have to re-stripe the tape, and it can be very difficult to get the new stripe to start at exactly the same place as the old stripe. If

There's a more sophisticated way of putting timing information on a tape, and that's to use SMPTE timecode. SMPTE always runs at the same speed, and there is no tempo information embedded in it. For it to drive a MIDI sequencer, you need a device that will convert the SMPTE information into MIDI clocks and pointers. This device is called (not surprisingly) a "SMPTE-to-MIDI converter".

To use such a device, you first tell it the SMPTE time — the hour, minute, second, and frame — at which you want the sequence to start, by entering the numbers on a numeric keypad. Then you tell it how many measures long the sequence is, and at what tempo and meter (time signature) it should play. If you want to change the tempo or meter in the middle of a sequence, you have to specify at what measure and beat the change occurs, and what tempo or meter to change to. This list of changes is known as a "tempo map". If you want to make any changes in an existing tempo map, you have to do them one at a time.

The converter then reads the incoming SMPTE timecode from tape. When it encounters the SMPTE number you've programmed as the start time (also known as the "offset"), it sends out a Start command, and follows it with Timing Clocks at a rate corresponding to the tempo you've programmed (= tempo in quarter-notes per minute x 24). The sequencer starts, and then follows along in step.

If you start the tape later, the convertor reads the first SMPTE number it encounters, and calculates the bar and beat of the sequence that that number corresponds to, using the offset time and the tempo information in its tempo map to do the arithmetic. It then sends out a Song Position Pointer to the sequencer, followed by a Continue, and then Timing Clocks at the appropriate tempo. Because the convertor knows the tempo at every point in the piece, it can send out correct Song Position Pointers regardless of where you start the tape.

SMPTE-to-MIDI-Clocks-and-Pointers conversion was the standard for the first couple of years of MIDI, but it has some disadvantages. Any tempo changes that are part of a

cha... into ... nver... ne a... me ... nd ... very tedious, especially when you are dealing with music that speeds up or slows down a lot, or with film music in which cues have to be cut very precisely to visual events (and in which last-minute timing changes are all too common). Long, smooth accelerandos or ritards can be equally onerous. Experimenting with different tempos to see how things fit is difficult and even dangerous (you may not be able to recover your original data if you make a mistake). If a section of a sequence has to be cut out, or a new section has to be added, often the entire tempo map has to be reconstructed from scratch.

Also, most hardware convertors can store only one tempo map (although some can store two or three), and so every time you want to work on a different sequence you have to enter all of its tempo changes by hand. Some convertors allow off-loading entire tempo maps using system-exclusive codes, which makes things a little easier but can still be unwieldy.

SEARCHING FOR ALTERNATIVES

Engineers looking at this situation wondered if there was a way to bypass the hardware-based tempo-map convertor, and get a sequencer to respond directly to SMPTE timecode. Most sequencers already have built-in tempo maps, and having to duplicate that in hardware seems so unnecessary. But SMPTE is analog information, and computers need to deal with incoming data in digital form.

One of the first methods to get SMPTE into a computer was devised by Mark of the Unicorn and Southworth Music Systems, two Macintosh developers, who collaborated on "Direct Time Lock". In this system, a SMPTE number coming into a special conversion device first triggers a system-exclusive command. This command contains the initial SMPTE number, and it is sent to the sequencer, which calculates a starting point from it, based on its own internal tempo map. Then MIDI Timing Clocks are sent at a *constant* (non-tempo-dependent) rate of one per frame (approximately 30 per second), and the sequencer uses

... digital signals, are used by ... 's ... hand... and ... Lab's ... SMPTE readers ... Atari ST family of computers.

One problem with Direct Time Lock is that once the sequence starts, the sequencer is on its own — if there is a tape dropout, or a timing byte gets lost or scrambled, the sequencer will go out of sync with the tape, and will have no way of catching up, or will simply stop. Another is that the timing resolution is low (even lower than Clocks and Pointers, except at very slow tempos), which means if the tape's speed is not absolutely constant, the sequencer can drift in and out of sync. (A new version of Direct Time Lock, called "Direct Time Lock Enhanced", sends clocks four times per frame.) The fact that Direct Time Lock is not part of the MIDI spec, but instead uses Southworth's System Exclusive codes (and they are now out of business), means that other manufacturers would be extremely reluctant to implement it, although some have, to make sure their hardware works with Mark of the Unicorn's Performer.

MTC TO THE RESCUE

MIDI Time Code addresses all of these problems, and the problems of MIDI Sync too. The work primarily of Evan Brooks of Digidesign and Chris Meyer, then of Sequential Circuits, MIDI Time Code uses previously unassigned bytes in the MIDI spec, and "Universal System Exclusive" MIDI commands. It took more than a year of debate within the MIDI Manufacturers Association and the Japan MIDI Standards Committee, but their proposal was finally incorporated into the MIDI spec in 1987.

MIDI Time Code consists of two types of messages: Real Time and Cueing. Real Time messages are the ones in use today — Cueing messages have yet to be implemented in any existing products (the Sequential Studio 440 was a first attempt at using them), and we'll talk about them later.

Like Direct Time Lock, there are two types of Real Time messages: Full and Quarter Frame. The Full message, which consists of ten bytes, is sent when a tape starts, stops, or repositions itself. As its name implies, it contains the full

secc ... 30-fr... vide...

The Quarter Frame message, which consists of two bytes, serves as the timing reference for the receiving device to base its tempo map on, but it has another function as well: it conveys the SMPTE number, so the receiving device always knows where it is supposed to be. But it doesn't do it all at once: it breaks down the SMPTE number into "nibbles", or pieces of bytes (I didn't make this up), and the second byte of each Quarter Frame message is one nibble. Each nibble contains one-eighth of the SMPTE number, so when the receiving device has read eight of these messages (which takes two frames), it can reconstruct the entire SMPTE number.

This means that the timing resolution of MIDI Time Code is four times better than Direct Time Lock, and also better than MIDI Sync for tempos up to 300 beats per minute. Therefore, lockup with a moving tape is very good, even if the tape speed varies. In addition, in the event of lost data, a receiving device can check and correct ("chase") its position as often as 15 times each second.

Converting SMPTE to MIDI Time Code is a relatively straightforward process. There is no need for such a convertor to have any controls at all: SMPTE comes in, MIDI Time Code goes out. The MTC then goes over an ordinary MIDI cable to the sequencer or computer interface (many MTC convertors are built into computer/MIDI interfaces, saving a step). Because they have no buttons and displays, and need no internal RAM for tempo maps, SMPTE-to-MTC convertors are considerably less expensive to make than SMPTE-to-Clocks-and-Pointers convertors.

MIDI Time Code-to-SMPTE convertors are also possible, except there are none in existence yet that I am aware of.

OW IT'S USED

The most immediate advantage to using MTC with a sequencer is that the starting time and tempo information

s... to ... late... the ... wh... measu... ...at to start at, and what tempo to run at.

This has several major benefits. First, setting up tempo changes in a well-designed computer sequencer is much easier than doing it on a hardware convertor box with a keypad and a single-line alphanumeric display. Most good sequencers let you do useful tricks like create smooth accelerandos and ritards, or design a complex tempo and beat pattern that runs over several bars and have it repeat many times. Some sequencers even let you run different tracks at different tempos. When using MIDI Time Code, you can continue to use all these features, letting the computer worry about making sure the numbers are exactly right.

Consider what happens when a film director gives you a new version of a scene in which two seconds have been cut. If you are using conventional MIDI Sync, you have to calculate the new time, look in a "click book" to see what the tempo should be to make the existing music fit the new time, and then re-program the convertor. If there are multiple tempo changes within the section, you have to recalculate and re-program all of them, one at a time, and hope that they come out all right. With MIDI Time Code however, and a sequencer with a "Fit Time" feature, you can just select the music that is to be re-timed, and tell the sequencer to recalculate its internal tempo map to reflect the new length. (Fig. 1)

Another advantage of MIDI Time Code is that because the sequencer's internal tempo map is always being used, you can listen to the sequence *without* the tape running and hear exactly the same thing you would *with* the tape running. When you're editing a sequence, it is usually much easier just to use the computer's locating functions to go back and check what you're doing, than it is to rewind the tape, roll it, and wait for the synchronizer to catch up each time.

A third advantage is that a MIDI Time Code system can easily recover from a momentary loss of sync. If there is a SMPTE dropout, lasting anywhere from a single bit to a couple of seconds, many MTC convertors have the ability

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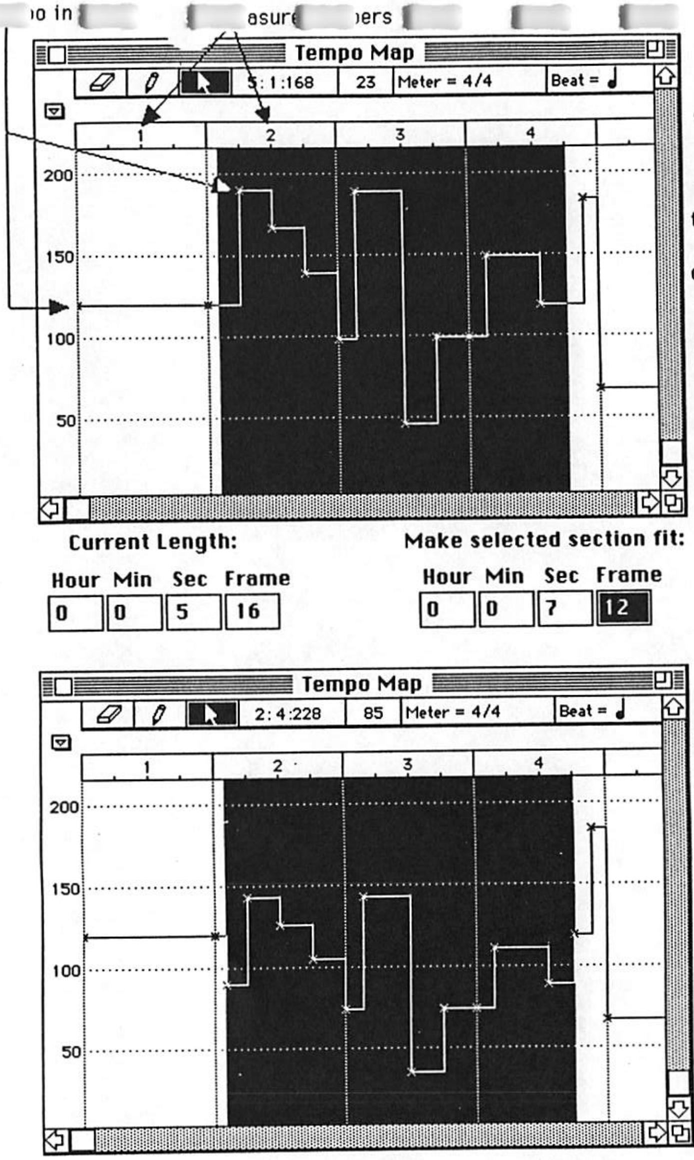

Using a "Fit Time" function on a sequencer's internal tempo map to change the overall length of a segment that already contains multiple tempo changes. This feature can only work if the sequencer is locked to MIDI Time Code, not conventional MIDI Sync (Clocks). [example is from Passport Designs' Pro4 for the Macintosh]

to "flywheel", or continue to generate MTC for a period of time until the SMPTE resumes. (Some software sequencers also have flywheeling capability built in as well.) When the SMPTE starts coming in again, if there is any discrepancy between where the sequencer is and where it's supposed to be, the sequencer can speed up or slow down to match the SMPTE, like a tape deck that is chasing code from another deck. This chasing ability also means that when several devices reading MIDI Time Code are running simulta-

…ously, they will always stay in sync, ev… there are dropouts anywhere in the system.

Finally, MIDI Time Code allows complete slaving of a sequencer: there is no need to place a sequencer into "Play" mode every time you want it to start, as you have to with some other forms of sync. Most sequencers, when they read valid Quarter Frame messages, will start automatically. This may seem trivial at first glance, but it can prove to be a major convenience when punching tracks in on a multitrack tape, or doing film scoring and effects, when you are constantly stopping, rewinding, and starting tape. With MTC, the sequencer can even be started *after* tape starts rolling, which is impossible otherwise.

It's worth noting that MIDI Time Code won't replace MIDI Sync in every situation, but will often work alongside it. For example, a drum machine that only reads MIDI Sync can still be used in a MIDI Time Code studio: a sequencer reading the MTC can itself generate MIDI Clocks and Pointers to drive the drum machine *while it's running*, and the sequencer's tempo map will take care of tempo changes for the drum machine.

THINGS TO WATCH OUT FOR

There's no doubt that MIDI Time Code is wonderful stuff, but there are some things about it that the user must be careful of. The most talked-about of these is the amount of MIDI bandwidth the commands take up. It's well known that if a MIDI line is asked to carry too much data it slows down, and the music coming out of it can sound jerky and out of sync. Does MIDI Time Code contribute to this problem?

Theoretically, MTC within a MIDI stream uses up between eight and 12 percent of the available bandwidth. If a MIDI signal is being pushed to its limits in terms of timing, adding MTC on top of it could put it over. But this is not really the issue — if you're running a MIDI line that close to the edge, you probably will have problems whether MTC is there or not. A more serious issue is the converse: how conventional MIDI interferes with MTC.

es r... g MIDI Time Code will generally be more sensitive to timing information, both because they are constantly calculating tempos and not just following Clocks, and because MIDI Time Code has a relatively high resolution. Because an MTC Quarter-Frame message is two bytes long (as opposed to a Clock's single byte), and because a MIDI data stream can be full of other messages of varying lengths (including System Exclusive messages, which can be very long indeed), the MTC message can get pushed around slightly. This results in a form of digital "jitter", and can have a small but noticeable effect on the timing of any device slaved to the MTC data.

For this reason, it's considered good practice to have MIDI Time Code on its own MIDI line, separate from conventional MIDI data. In fact at least one sequencer program *forces* you to keep the streams separate. When it is used this way, MIDI Time Code's resolution is as good as the clock rate of the device being used to generate it: a computer running at 1 MHz should be easily able to generate MTC with a resolution of 2 microseconds (two clock cycles), which is well within the SMPTE specification's tolerance for the initial address bit, which is +/- 10 µsec.

This means that MTC could be used, theoretically, to synchronize a tape deck. However, there are currently no decks on the market that have this facility. There are a few tape decks that are "MIDI compatible", but the extent of this compatibility is that the transport controls can be remotely operated using MIDI System Exclusive commands that can be generated by a computer or sequencer. The timing master for all of these decks remains SMPTE timecode, striped on an audio track.

Not every device that performs SMPTE-to-MTC conversion does it in quite the same way. A number of convertors just send Quarter Frame messages, and never send Full messages. This is no big deal if you are only using MTC to lock to a moving tape striped with conventional linear-recorded SMPTE timecode, but it can run into problems if you are using VITC.

Since VITC can be read while tape is standing still or moving at a wide range of speeds, forwards or backwards, it can be used in the MIDI studio to allow a composer to find hit points for music and effects in a video by rocking the ... tape ... and the ... controlling ... computer ... the number of the frame it sees. This is impossible unless the convertor sends full MTC messages.

It's also possible to misinterpret MIDI Time Code. If you think about it for a moment, you'll realize that MTC is always running behind. Since it takes two frames (eight Quarter Frame messages) to send a complete SMPTE number, by the time the number is received, it's already two frames later. This constant offset should be built into any software that reads MTC, or else nothing will ever line up right. However, there were reports that the designers of one early MTC-compatible program forgot this, and things were pretty weird for some of their users for a while.

Even if it's reading the code correctly, there can be differences in the way programs deal with MTC. An MTC-based sequencer can catch up with lost data, but again, not every sequencer does it exactly the same way. Just as different tape synchronizers behave differently when they are dealing with off-speed tape decks, different sequencers use different algorithms to chase MTC. Some may check the code every ten frames, while some may check it every beat, and others may only check it once each measure. If the algorithm is designed well, its action should be inaudible, but if you are running two MTC-based programs together that use very different algorithms, you may actually hear them go in and out of sync.

BEYOND SEQUENCING

The best part about MIDI Time Code is that its potential usefulness extends far beyond just locking sequencers to tape. It's an effective and inexpensive way to get SMPTE numbers into a computer for any purpose. Digidesign's *Q-Sheet* event-list editor for sound effects and studio automation couldn't run without it. *Q-Sheet* is really a sequencer that uses SMPTE numbers instead of bars and beats, and when it is running from MTC, you can look at a video that shows the flash coming from a gun barrel at 2 minutes, 45 seconds, and 4 frames, and type that number in along with a command for a sampler to play a gunshot at that exact moment. With a sequencer that only reads

conventional MIDI, you would have to set your convertor to 4/4 at 120 bpm, and then calculate that the gunshot should occur on measure 82, beat 3, clock 60. Imagine doing that for every effect in a two-hour film!

Hit-list generators like Opcode's *Cue* and Passport's *Clicktracks* rely on MIDI Time Code to take a huge amount of the drudgery out of film scoring. As the composer watches the video, and as MTC comes into the computer, he hits the spacebar every time he wants to mark a hit. The hits are recorded, referenced to their MTC times, and the program can then devise a tempo map that will put all of them on significant musical beats. The tempo map can be saved as a standard MIDI File, and be transferred to a sequencing program, which also locks to MTC. So while the hit-list generator arranges all the timings for the music, and all that remains for the sequencer to do is fill in the notes.

Sound Tools, the hardware and software package from Digidesign that turns the Macintosh into a digital audio workstation, depends on MIDI Time Code for communicating with the outside world. When the system is told to record a section of an incoming audio signal, like a line of dialog or a location-recorded effect, it knows where to start and stop recording by the section's MIDI Time Code numbers. When it plays back a track while slaved to an audio or video tape deck, that slaving is accomplished through MTC.

Apple's MIDI Manager, a musical multitasking system that is catching on with Macintosh-based studios, also depends on MTC. The system can run sequencers, effects-list editors, algorithmic composers, and direct digital playback from RAM and/or hard disk all at the same time, and the common link for all of these programs is MIDI Time Code.

THE FUTURE

Just as MIDI itself has gone far beyond what its developers imagined, so MIDI Time Code has the potential to take MIDI technology into areas quite removed from the synthesis and sequencing worlds in which it has become so familiar.

First of all, MIDI Time Code can be an integral part of an extensive MIDI-based studio automation network. As we've seen, a tape transport can be controlled from MIDI, and extrapolating on this idea, one can foresee a single central control surface — computer, mixing console, or a dedicated controller — handling everything in the studio, eliminating the constant running back and forth between tape deck and sequencer that's part of many MIDI studios today.

Then there are the Cueing messages, the "unknown" half of the MIDI Time Code spec. Cueing messages allow MIDI to become a kind of "Device Control Language", in which whole sets of complex time-based commands can be sent to various devices around a studio *before* tape is rolling. These commands are stored in the devices' memories, and tell them that when the tape starts to roll and they receive a certain Time Code number, they should execute a certain operation.

There is enough space in the MTC Cueing command set to accommodate all possible functions that any studio device might perform. Every device would respond to the command set in the same way: the same "Play" command can be recognized by a video deck, a CD player, a sequencer, or even a turntable.

A very simple example would be pre-programming a multitrack tape deck to do multiple punch-ins and -outs at different times on different tracks in one pass. Before you start the music, you send a set of Cueing commands to the deck that specify all the punch-in and -out times. Then when you start the music, the deck only has to receive timecode, not any actual commands, and it executes the punches at the programmed times.

With SMPTE-based console automation systems one must program each individual device to do its thing at the right SMPTE time, and send SMPTE to all of them. With MTC's Cueing commands, on the other hand, you can program everything from a central controller — your computer — and make changes (like that film scene that's now two seconds shorter) and store files with ease, like a CMX video editor, but infinitely more flexible. You can send both SMPTE and MIDI Time Code around the room in parallel, routing each wherever it is needed.

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ueing messages can't be used to synchronize and a mate video systems, lighting systems, theatrical evei multimedia, museum displays, robots, interactive ad ing displays, or anything at all that can be controlled digitally and relates to time.

What MIDI Time Code needs now is more people to recognize its potential. Manufacturers need to stretch their imaginations a bit and look at all the things that can be done with it, and users need to learn and take advantage of what's already available, and then put pressure on manufacturers to do more. It's an idea whose time, if you'll excuse the expression, has come.

The author is a composer, engineer, consultant, and teacher in the Boston area. He is a contributing editor to Recording Engineering Production magazine, and writes frequently for numerous trade and consumer publications in the US and Europe.

ANSI/SMPTE 12
Revision and Rede
ANSI V9

American National Standa

for televisio time and control cod video and audio tape 525-line/60-field syste

Approved January 29, 1986

Sponsor: Society of Motion Picture and Television En

ANSI/SMPTE 12M-1986

1. Scope

1.1 The first part of this standard specifies a format and modulation method for a digital code to be recorded on a longitudinal track of video and audio magnetic tape recorders. The code is to be used for timing and control purposes.

1.2 The second part specifies the digital format to be inserted into the television signal vertical interval to be used for timing and control purposes in video magnetic tape recorders. This part also specifies the location of the code within the television baseband signal and its relationship to other components of the television signal and to the longitudinal track code described in the first part of this standard.

2. Referenced Standards

This standard is intended for use in conjunction with the following standards:

EIA Industrial Electronics Tentative Standard No. 1, Color Television Studio Picture Line Amplifier Output Drawing

International Standard ISO 646-1983, Information Processing — ISO 7-Bit Coded Character Set for Information Interchange

International Standard ISO 2022-1982, Information Processing — ISO 7-Bit and 8-Bit Coded Character Sets — Code Extension Techniques

3. Longitudinal Track Application

3.1 Modulation Method. The modulation metho shall be such that a transition occurs at the be ginning of every bit period. "One" is represente by a second transition one half a bit period fron the start of the bit. "Zero" is represented whe there is no transition within the bit period. (Se Fig. 1.)

3.2 Code Format

3.2.1 Frame Make-up. Each television fram shall be identified by a unique and complete ad dress. A frame consists of two television fields o 525 horizontal lines. The frames shall be numbered successively 0 through 29, except as noted in 5.2.2 (Drop Frame). If color frame identification in the code is required, the even units of frame numbers shall identify Frame A and odd units of frame numbers shall identify Frame B, as defined by EIA Tentative Standard No. 1.

3.2.2 Frame Address. Each address shall consist of 80 bits numbered 0 through 79.

3.2.2.1 Boundaries of Address. The address shall start at the clock edge before the first address bit (bit 0). The bits shall be evenly spaced throughout the address period, and shall occupy fully the address period which is one frame. Consequently, the bit rate shall be 80 times the frame rate in frames per second. (See 3.2.1 for definition of a television frame.)

ansi American National Standards Institute, 1430 Broadway, New York, N.Y. 10018

ANSI7...

Digital

"involving or using numerical digits expressed in a scale of notation to represent discretely all variables occurring in a problem"

Boolean algebra- a deductive logical system, usually applied to classes, in which, under the operation of intersection and sysmmetric difference, classes are treated as algebraic quantities.

Until this point, all video systems represented light information (brightness, saturation, and hue) as an *analog* electrical wave form. Bright light would be represented as a high voltage on a wire and a lack of light would be a low voltage. The strength of the light is transduced to the strength of the electrical signal.

Let's take the case of light represented as electrical information. In an analog system, the brightness value of light is converted in each moment of time to an electrical value. Comparing the electrical signal over time with the brightness values over time will reveal two graphs of similar shapes. The shape of the signal over time will be the same.

An analog system will process signals via electronic circuits using capacitors, resistors, and transformers. A digital system processes signals by using a computer.

Preparing information for a computer
A computer can process only one type of information: binary, digital signals. In other words, a stream of bits (either on or off) will carry the same information that an analog waveform carries. How is an analog waveform prepared for use by a digital computer?

Turning analog to digital (analog-to-digital converters, A-D)
An analog waveform is simply an electric current that varies in strength (and voltage) over time. As closely as one looks at an analog waveform, there is always a measurement of time and of voltage height (amplitude) which is further divisible. This means there are no discrete values of voltage or time in an analog system. There is simply a constant transduction of one energy type to another, and the information is continuously being represented.

A digital system must break signals down into the tiny consituent values that a computer can process. A computer's basic data requirements are either a zero or a one, a low voltage or a high voltage. Again, how can a continuous signal be translated into discrete zero's and one's?

Quantizing, measuring

For every moment of time, a waveform is at some value. At one moment of time, a measurement of the amplitude (height or voltage) of the waveform can be made. Remember that in a video system, the amplitude originally represented brighness values and it was transduced to an analog electrical signal. Each measurement (or sample) of the waveform can be represented as a 0 or a 1. If the voltage is high, the sample is 1, and vice-versa.

This measurement system is probably too discrete. In other words, there are a whole range of subtle values between the lowest voltage (lowest brightness) and highest voltage that are not represented in this one-bit (0 or 1) system. Adding a second bit means four brightness or voltage values are possible to represent: 00, 01, 10, 11. Now there are four discrete brightness or voltage values that the computer can process.

The way to increase the subtlety of the system is to increase the number of bits per sample that represent the brightness or voltage level. 10 bits can represent 1024 gradations from the darkest dark to the brightest bright. Early digital systems used 8 bits to represent 256 brightness values. Newer systems use 10 bits.

The number of bits chosen to represent the gradations of level per measurement is known as the *quantization level.*

Sampling

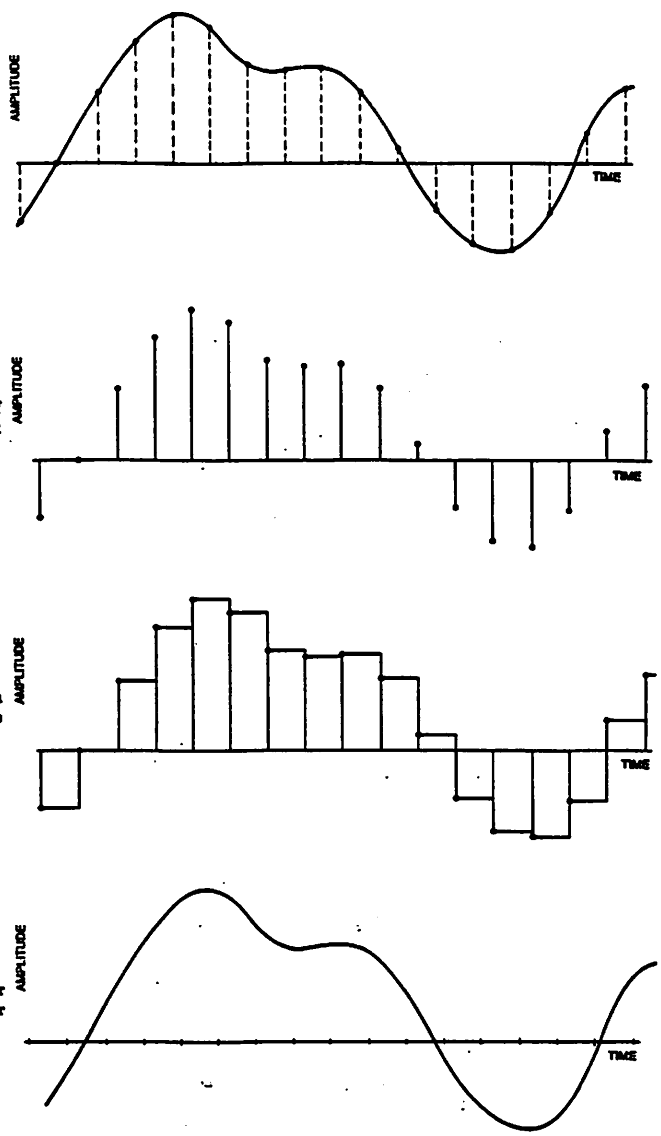
Now that the waveform is being measured and the strength of the waveform is translated to be a binary number, the question becomes: how often should the waveform be measured?

The answer is simple. Consider what the highest frequency in the system is and sample *twice as often.* This is known as the Nyquist Theorem.

For example, in a digital audio system, the highest frequency necessary is the highest frequency a human can perceive, which is 20,000 Hz. Therefore, the number of measurments to be taken in one second (the sampling rate) must be 40,000 times per second or 40,000 Hz. By measuring twice as often as the highest frequency, you are ensuring that the system will accurately reconstruct the information. To see how this works, imagine a waveform at a frequency of 20,000 Hz. This means one cycle happens in 1/20,000th of a second. If the system takes measurements of the waveform (sampled) 1 time every second, 20,000 cycles would not be accounted for every second. If 10,000 measurements (samples) are made, half of the cycles would be accounted for and the other half would be missed. Missing cycles of the signal means that the computer will be processing inaccurate information *and* the waveform cannot be accurately reproduced for the given information.

SAMPLING

In digital recording, the analog sound wave is "digitized" by sampling the instantaneous signal amplitude at very short intervals of time. These instantaneous samples are turned into binary digits, or bits, that can be understood and manipulated by the computer.



(A) The input analog signal is sampled.

(B) The numerical values of these samples are stored (effect of quantization not shown).

(C) Samples are held to form a staircase representation of the signal.

(D) An output low-pass filter smooths the staircase to recover the original waveform.

Fig. 2-2. Discrete time sampling: a band limited signal can be sampled and reconstructed without loss.

It's really a simple game of connect-the-dots. The less dots (samples of the waveform amplitude), the less precise will be the reproduction of the original signal. Constellations in the night sky may be made up of ten or fifteen stars, yet the pictures they represent are full of curves and details. If a computer was given the task of drawing Ursa Major (the Big Bear) from the coordinates which represent it, there is a very slim chance that it would reproduce all of the subtle curves and features of a bear. More likely, it draw a simple, angled line. The Big Dipper, as Ursa Major is also called, is what the computer might output.

With this constellation analogy in mind, imagine a picture of a bear. To accurately represent all of the features of the bear, a lot of points must be measured on the picture if it is to be accurately reproduced. If only one point is chosen for the head, two more for each arm, and two for the feet, only a stick figure will result at best. The more sample points the better, up to a point. There is no need to sample more than the final product necissitates. In an audio system, for example, sampling more than twice the higest frequency (20,000 Hz) would capture information that exists (ultrasound frequencies) but that humans would never hear.

Analog-to-digital converters

Any device which takes an analog signal and prepares it for computer processing is known as an analog-to-digital computer. All signals originate as analog signals (or, for practical purposes, infinite resolution signals, in which there is always another level of divisibility in which to measure). The computer requires the quantized, discrete digital signals via binary codes. Different information is represented by different digital codes.

Audio stored on a compact disc is sampled 44,100 times per second (44.1 KHz). Each sample is a some discrete value between 0 and 65,535, represented by 16 bits.

The CCIR 601 Digital Video standard defined COMPONENT DIGITAL VIDEO to be sampled at the following rates:

luminance channel, Y - 13.5 MHz (13.5 million samples per second)
color difference, R-Y - 6.75 MHz (6.75 million samples per second)
color difference, B-Y - 6.75 MHz (6.75 million samples per second)

Each sample has a value of eight bits (256 possible values between 0 and 255).

Once the analog signal has been coded as digital information, the fun begins.

Digital signal processing

Digital signal processors (DSP's) are computers that perform specific operations on converted analog signals. For example, audio digital signal processors are built to simulate effects like reverb, flanging, chorusing, or equalization. All of the processing is mathematical operations on the numerical values which represent the original analog waveform. Because there are so

many samples per second, digital signal processors are doing vast amounts of calculations per second to keep up with the throughput of data.

CHAPTER 16: THE DIGITAL DOMAIN AND BEYOND

The title of this chapter may be an anachronism to some. After all, the digital domain *is* the likely 'final frontier' for audio, at least in terms of how audio is processed and recorded. It won't, however, signal the evolutionary end of recording technology. Rather, it will open doors to endless possibilities. We've seen how dramatically the Compact Disc has changed the reproduction of music. The changes that the digital age will bring to recording are even greater.

In the late-1970s, the first multi-track digital tape recorders began to appear in professional studios. Many observers, citing the meteoric advances made by the computer and electronics industries through the '70s, expected that within a decade, digital recording would be the only game in town. After all, digital recording was initially expensive, but its sonic advantages of no tape hiss or distortion and crystal-clear reproduction were certain to create a fantastic demand for more affordable digital recorders.

A decade later, however—while digital multi-track recorders have proved to be *de rigueur* for any self-proclaiming 'world-class' studio—multi-track *analog* tape recorders also remain popular, even in many professional studios.

A number of factors have led to the delay of the 'digital revolution.' For one, the first digital recorders faced a lot of opposition from seasoned producers and engineers. These people were rather comfortable with analog recorders, and many were accustomed to the slight but audible degrees of distortion they induced—to the point where digital recorders sounded 'cold.' On top of this, some early digital systems did suffer from their own peculiar sonic quirks, though modern systems are generally considered sonically excellent.

There have been other important delay factors, as well. Analog recording has made great leaps since the '70s. Dolby and dbx noise reduction systems have helped reduce tape noise in many professional and personal studios. Four-track cassette multi-trackers, 1/4" 8- and 1/2" 16-track open-reel machines, and more, are innovative formats that work very well in many applications. And Dolby Labs, with its amazing SR noise reduction (page 61), might like us to think that there's no reason for anyone to need a digital recorder.

Along with all of this, both manufacturers and consumers have been spending their respective dollars (or yen) developing and buying such amazing tools as:

- Digital sequencers, drum machines, samplers, and other MIDI-based gear.
- Affordable digital signal processing, such as delays and reverbs.
- Personal computers, to handle many studio chores.

So in fact, the digital revolution *has* been taking place, and the first digital products to reach most studios have been signal processors, MIDI sequencers, digital mixers—such as the Yamaha DMP7—and other such gear. Furthermore, a sequencer is functionally a multi-track digital recorder for MIDI instruments. Samplers are also digital recorders, though they may be limited in terms of time and tracks.

Let's find out more about digital audio—how it's works, what to consider when evaluating digital gear, and some of the options it will bring to the personal and professional studios of the future.

Digital Audio Basics

Whether it's a groove cut in a wax cylinder or magnetic particles rearranged on tape, analog recording works by creating a reproducible 'model' of the sound being recorded. Similarly, all analog audio gear uses varying degrees of electricity to process and transmit audio signals. Whatever the medium—wax, magnetic particles, or electricity—analog audio alters it in such a way that it is directly analogous to the source.

Digital audio—and this applies to digital recording, signal processing, or whatever—works very differently: It analyzes sound and describes it in terms of numbers. These numbers can then be stored as a digital code, altered for level and signal processing, and read back—for ultimate playback.

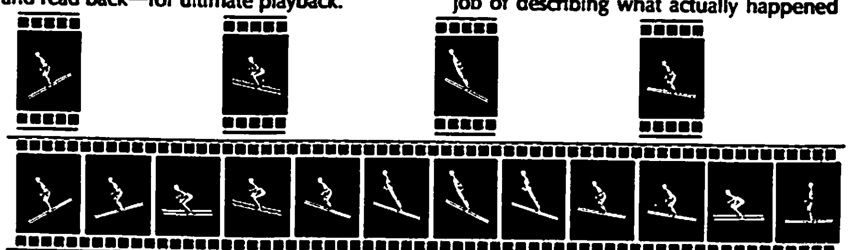


Figure 16.1 'Sampling' a ski jump every three seconds (top) doesn't show us the same story that a 'sampling rate' of every second (bottom) shows us. In a similar way, the sampling rate at which a sound is converted to digital audio has a direct bearing on the quality of audio that we hear.

Since only numbers are stored, altered, and played back, anything that isn't a recognizable number is ignored. For example, digital recordings, even those made with magnetic tape, suffer no tape noise. That's because tape hiss is random magnetic particles, that can be read back by *analog* tape heads (which are looking for any magnetic variations), but mean nothing to a digital recorder's heads.

Sound is converted to digital numbers by taking numeric 'snapshots,' or *samples*, of the sound many thousands of times a second. A digital numeric code is assigned to each sample; as the sound changes, so do the codes. Each sample, incidentally, is like a pulse which registers a value for the audio being sampled. For this reason, digital recording is sometimes known as *PCM* recording, for *Pulse Code Modulation*.

The codes are very important, since they are the digital representations of the sound. This brings us to the next point: Just because something is 'digital' is no guarantee that it's the epitome of audio perfection. After all, there are digital sampling instruments which have poorer frequency response and noise specs than the average 4-track analog mini-studio. More than anything else, a digital audio device's fidelity depends on two factors:

- How many times per second it takes a 'digital snapshot' of the incoming audio signal; this is known as the *sampling rate*.
- How detailed each 'snapshot' is; this is known as the device's *resolution*, and is expressed in *bits*.

Let's learn more about these two.

Sampling Rate. Imagine you and a friend are sports photographers, assigned to cover a ski jumping event. While your friend's camera is armed with a motor drive, capable of taking one frame a second, your drive has frozen up—leaving you with a manually-wound camera.

After the jump, the two of you thaw out by developing your film. As the images appear on the contact sheets, you compare each others work. Of course, the results are almost predictable: In the 12 seconds or so it took for the gold medalist to complete her jump, you managed to take just four photos, whereas your friend squeezed off a succession of 12 motor-driven shots (Figure 16.1). Clearly, your friend's shots do a much better job of describing what actually happened throughout the winning jump.

And so it's the same story with digital audio. As we know, incoming audio must be sampled—like a series of audio photographs—'x' number of times per second in order to be converted to digital information (Figure 16.2). The more samples per second, the better the description of the incoming audio. What sort of sampling rate is necessary for quality audio?

Well, it may be obvious that a sampling rate of 1 Hz—one sample per second—would be fairly ludicrous. Just as with our ski jump photo shoot, such a system wouldn't 'fill in the blanks,' and would be useless for any musical application. So that tells us, first of all, that we need to sample at a rate fast enough to digitize any possible musical changes in rhythm, level, notes, and so forth. Perhaps a couple thousand times per second would be more than adequate, right?

Wrong. We haven't taken into account the frequency of the sound we're recording. Remember, humans can hear sounds up to 20 kHz. If a cymbal is shimmering at 16 kHz, it's oscillating at 16,000 times per second—

and a sampling rate of 2,000 times per second would simply be unable to create a proper digital 'picture' of that sound. As we ···n see in Figure 16.3, if we sample a frequency with too low a sampling rate, we ...id up with a very odd picture of that frequency. This is known as *aliasing*, and sounds like weird, random pitches. The solution to aliasing is to use a lowpass filter (page 45) to remove any excessively high frequencies.

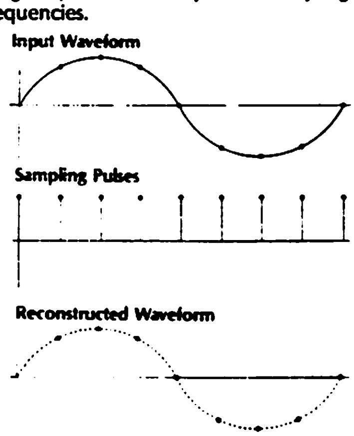


Figure 16.2 An appropriate sampling rate for this frequency. As we can tell, the digital sample is a fine reconstruction of the original waveform.

This leads us to the next logical assumption: The sampling rate needs to be the same frequency as the highest sounds we want to record, right?

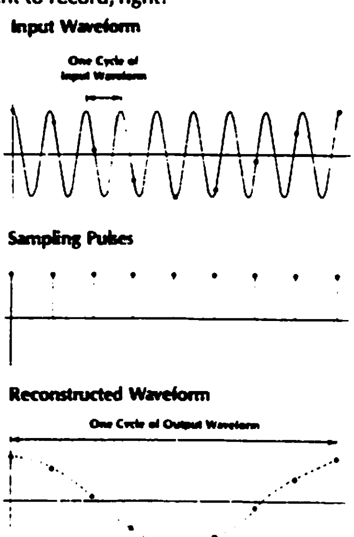


Figure 16.3 This sampling rate is far too infrequent to reconstruct the incoming waveform. Without a high enough sampling rate, the result is known as aliasing, as shown.

Logical, but wrong again. In fact, the sampling rate needs to be at least *twice* as high as the highest audio frequency we wish to record. Figure 16.4 shows an audio signal being sampled: what we see is actually the highest frequency audio signal that we can digitize with the sampling rate shown. If we ··mpled the audio signal at the *same* sam- g rate as its frequency, we would only be ..npling each peak, as indicated. With a sampling frequency of twice the audio signal, however, we're able to make a complete sample. The reasoning we've just de-

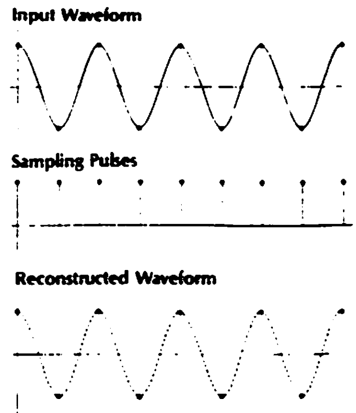


Figure 16.4 Given this sampling rate, this is the highest frequency that can be converted without suffering the dreaded aliasing. The sampling rate must be at least twice the frequency of the audio waveform's frequency, in order to reconstruct all peaks and troughs of the wave.

scribed is often referred to as the *Nyquist theorem*, incidentally, and tells us that if we want 'audiophile' performance of 20 kHz, we need to sample digitally at least 40,000 times per second.

Professional multi-track machines usually have switchable sampling rates of either 44.1 or 48 kHz. Compact Discs are produced with the 44.1 kHz sampling rate, and thus offer a top-end frequency response of 22,050 Hz. Many less-professional sampling instruments have sampling rates of 20 kHz or less, yielding usable—though not great—frequency responses of 10 kHz or less.

A high sampling rate takes up a relatively large amount of memory, which explains why all digital gear doesn't have sampling rates of 40 kHz or higher. In fact, since sampling instruments have limited memory, some allow you to select your own sampling rate. By lowering the rate, you lose high frequency response but gain extra recording time.

Bit-Rate Resolution. Sampling rate is only part of the picture when it comes to digital audio fidelity. To continue with our photographic analogies, let's consider film. Film is covered with light-sensitive particles. Fine-grain film has many more of these particles per frame than does a coarse grain film, and is thus able to take photos with higher resolution.

In digital audio, each sample is described in terms of digital numbers, known as *bits*. Simply, the more bits available to describe a sample, the greater the *audio resolution*.

As you may know, bits—either '1's or '0's—are the language of all things digital. When it comes to digital audio, the number of bits ascribed to each sample has a direct bearing on the dynamic range of the sample. For example, let's say we had a 2-bit digital recorder. That means each sample can have its level described, or *quantized*, to one of four *bit-word* permutations:

00	(Off)
01	(Louder)
10	(Louder still)
11	(Loudest)

As you can surmise from this limited range, a 2-bit system is pretty much useless for music. Much more useful are 8-bit systems, which gained popularity in the mid-1980s with several affordable sampling instruments, such as the E-mu Emulator II, the Ensoniq Mirage, and others. Eight-bit sampling, with bit words such as "01110001," offers 2^8 level possibilities, for a total of 256 steps. That degree of resolution offers a dynamic range of about 46 dB, which is okay, but not great. Multiplying the number of bits by six, incidentally, gives the approximate dynamic range.

For professional applications, 12-bit sampling is generally considered the minimum, with 4096 levels available, yielding a dynamic range of about 72 dB. While this might be fine for a sampling keyboard or digital reverb, it's still too low for quality digital recording. After all, any analog tape recorder equipped with Dolby C or dbx has a signal-to-noise ratio well above 70 dB.

In a 16-bit device, 65,536 levels can be described, which is equivalent to about 96 dB of dynamic range. With 16-bit digital processing and instruments now commonplace, professional-grade fidelity is both available and affordable.

As far as true digital recorders, there are several 18-and 20-bit systems available, offering extremely good dynamic range. Professional recorders by Mitsubishi, Otari, and others, offer switchable 16- or 20-bit performance. With a dynamic range of 120 dB, there's little need to exceed 20-bit performance.

Error Detection And Correction. When magnetic tape or disks are used as the digital storage medium, drop outs of the magnetic oxide can cause some big problems. In analog recording, a drop out will usually cause a momentary reduction or loss of signal. In digital recording, a drop out can cause all kinds of horrible 'glitches,' since the bits can be misread.

To counteract this, magnetic digital recorders include some form of automatic *error detection* and *correction* circuitry. Different devices use different systems, but the concepts behind all of them are to look for code inconsistencies in playback, and to create average values between the information preceding and following the error.

The ADA Chain. Figure 16.5 shows us a typical analog-to-digital-to-analog signal chain, commonly called an *ADA chain*. When an analog signal enters a digital device, one of the first things it encounters—after level matching switches, balancing amps, gain controls, and so forth—is a lowpass filter (sometimes called the *anti-aliasing filter*). As we just learned, the sampling rate of the device must be twice that of any audio frequencies it will encounter; otherwise, aliasing is possible. The filter removes the possibilities of aliasing.

Following that, the signal meets its moment of truth: the *analog-to-digital converter*, or *ADC*. At this point, the signal is sampled at whatever the 'local' sampling rate happens to be, and is resolved to the local bit rate. From this point on, we're no longer dealing with ordinary audio. Rather, it's just numbers, and we're free to do with them whatever we want.

For example, we can store them, to tape, disk, or in quickly-accessible *RAM* (*Random Access Memory*). That's the basis of digital recording, or for that matter, any sampling

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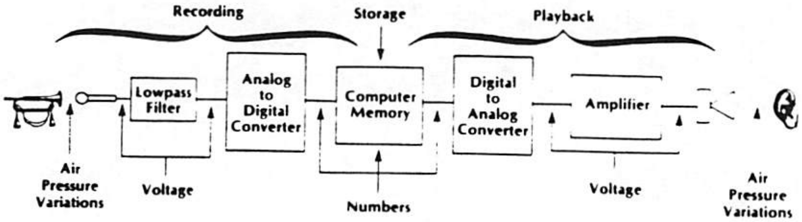


Figure 16.5 The basic components of a typical analog-to-digital-to-analog (ADA) chain. The lowpass filter removes those incoming high frequencies which may cause aliasing.

instrument. If we were designing a digital delay, on the other hand, we would design a circuit which would hold the numbers for user-variable amounts of time, and then spit them out. Similarly, if we were designing a digital reverb, we would create all kinds of interesting mathematical *algorithms*, to 'bounce' those numbers around the walls of simulated concert halls and caverns.

Whatever takes place, the beauty of *this* stage is that everything is numbers. There's no noise being added, since there's nothing in the circuitry which is susceptible to interference. This contrasts to analog gear—its circuits depend upon voltage changes, and noise can enter as stray voltage. Furthermore, digital manipulation offers many more options, in terms of what the sonic outcome will be.

Once the digital signal has been stored and/or manipulated, it can be either be passed along in digital form to a different digital device (as we'll see on page 128), or it can be retrieved for the return to analog. For this step a *digital-to-analog converter (DAC)* is used. The reconversion to analog remains a necessary step in order to pass the signal to another analog device, or to pass it to loudspeakers—after all, at this stage in human evolution we can't *hear* digitally!

Overload. A major difference between analog and digital gear is that digital gear can't accommodate an overloaded input signal. For instance, as you increase input levels, analog gear will gradually begin to distort. Digital gear will remain quiet until you actually overload—at that point the device has run out of numbers and will create some horrific-sounding distortion.

Consequently, many digital devices incorporate automatic limiters before their ADC stage.

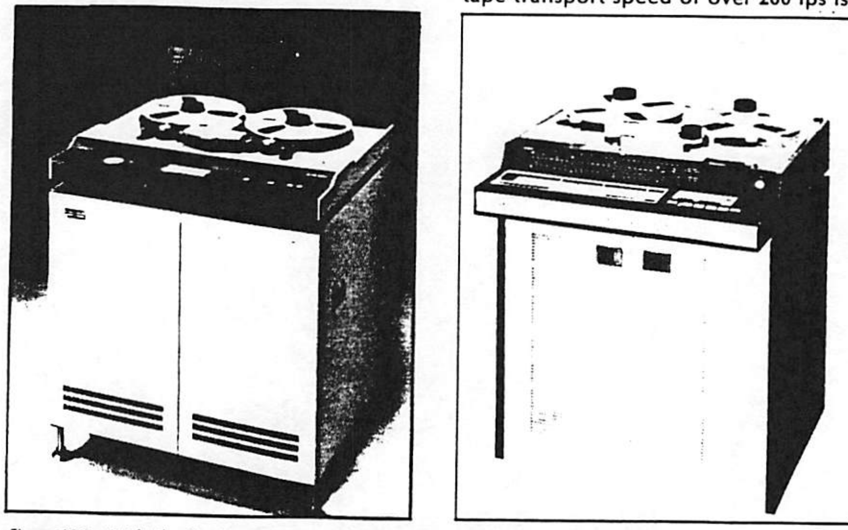


Figure 16.6a, 16.6b The Otari DTR-900 32-track digital tape recorder (left) is similar to the Mitsubishi X-850 32-track. The DTR-900 uses 1" tape and conforms to the 'PD' standard. The Sony PCM-3324 (right) uses 1/2" tape, and records 24 digital tracks in the 'DASH' format.

Digital Tape Recorders

Through the 1980s the digital multi-track recording market has been dominated by large, very expensive tape recorders, from the Sony *DASH* format, to the Mitsubishi *PD* format, and others. These machines all have 'fixed' headstacks, use 1" magnetic tape, and resemble analog machines in many ways.

To the chagrin of many professionals, DASH-format (*Digital Audio Stationary Head*) machines—by Sony, Studer, and Tascam—are incompatible with PD-format (*'Pro Digi'*) machines—by Mitsubishi, Otari, and AEG. For true 'world class' studios, this has often meant buying *both* machines, and has lead to hesitance from many potential digital buyers.

Of course, not all digital recording is multi-track. There are both DASH and PD 2-track machines, designed primarily for mixdown and editing. In the mid-'70s, the first digitally recorded analog LPs were produced with a Soundstream digital 2-track tape recorder, though that system was soon to lose ground to a less expensive type of digital 2-track: PCM-encoded video tape.

Figure 16.7 shows a PCM encoder system. Offered by Sony, Nakamichi, and others, PCM encoders receive a stereo analog line or microphone input, and put out a digitally-encoded signal, to be stored on a separate tape. Their digital output is, in fact, a *video* output, even though it contains no video information. Rather it's designed to be recorded on the video tracks of a video tape recorder (VTR), such as a 1/2" home consumer or 3/4" professional VTR.

When it's time for playback, the video output of the VTR is fed back into the PCM encoder, which in turn has a traditional analog output. Most encoders are switchable between 14- and 16-bit resolution, and usually record at the 44.1 kHz sampling rate.

PCM-encoded VTR recording offers a number of advantages over traditional fixed-head recorders. As we learned on page 116, VTRs use rotating heads. Since the tape is moving in one direction and the heads are rotating in another, an apparent tape transport speed of over 200 ips is achieved. All of this means a very stable transport in a small, inexpensive package—compared to the huge, expensive transports required by fixed-head recording.

Another variation on the rotating head theme is gaining ground, however, offering all the advantages of PCM-encoded VTRs and then some. The variation is *DAT* recording—initially known as *R-DAT*, for *Rotary-head Digital Audio Tape*—and we had our introduction to these machines back on page 29.

The DAT format has 16-bit resolution, and professional machines will offer sampling rates switchable between 48, 44.1, and 32 kHz. Though the current spec is 2-track only, some observers are confident that multi-track DATs are inevitable. Others suggest that the rotating head format of DAT

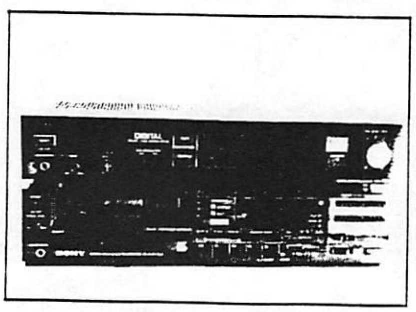


Figure 16.7 The Sony 501 PCM digital encoder, shown sitting on top of a Sony 1/2" Beta video deck. The 501 feeds a video signal to the video deck; encoded in this signal is the PCM audio. Upon playback, the deck's video outputs are routed back to the processor, and the signal is decoded into analog. The 501 operates at either 14- or 16-bit resolution.

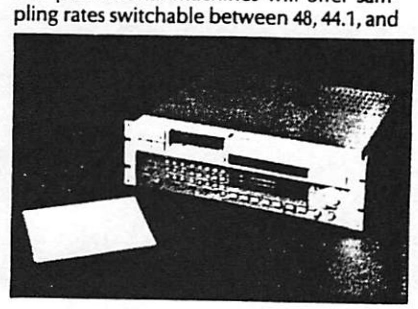


Figure 16.8 One of the first DAT recorders to be marketed, the Tascam DA-50. DAT recorders may very well be the mixdown format of choice through the 1990's.

Visual digital signal processing is also possible. Any digital data which represents the original hue, brightness and saturation (or R, G, B values) of a picture can be processed to alter a color, combine it with a second image source, or move it to a different location on the screen.

Digital video effects devices (DVE's) process video signals in real-time (immediate input to output with no waiting for processing). They are capable of altering image color, horizontal, vertical, and 3-D space posistioning on the screen, combining with other video signals, strobing (delaying the video signal) and bending the image in shapes according to mathematical processes.

In addition, all of the benefits of computer systems become accessible to video makers with digital systems. Images can be stored on hard disk instead of video tape, which means any frame can be accessed instantly. These systems are called nonlinear because no rewinding or fast-forwarding of a tape is necessary. Also, digital systems are as precise as their sampling rate and quantization level. Individual pixels can be isolated and adjusted if necessary. Perfectionism becomes a real possibility, however time consuming it might be.

Big, big numbers

Digital systems, for all of their advantages, must transfer huge amounts of data very quickly. CCIR 601 digital video has an approximate bit rate of 205 Mbit per second, or 205 million bits per second. This means there is a fluctuation of the electrical signal 205 million times per second. This is a huge amount of data and very precise technology is required to keep the transfer of this signal accurate. Recording this many fluctuations on a magentic tape is also a very precise game.

Most personal computers can't handle such a large amount of information and so the only way to fit the signal is to first *compress* unnecessary information to keep from overwhelming the CPU.

Compression

Compression is based on the notion that a lot of information transferred in a digital system is redundant, or can be considered redundant. By observing redundancies in a digital signal, the signal can be further coded using mathematically alogirithms to represent a large amount of data with less information. A simple example is to consider two hundred apples sitting in your kitchen at home. If every apple is identical, and you would like your friend to see what two hundred apples looks like sitting in *his* kitchen, you have two options. You can carry all two hundred apples from your kitchen to his, or you can bring one apple to his house and show it to him, making sure you mention there were two hundred to start with. It can then be your friend's responsibility to recreate the situation with 200 apples. In this way, you only had to carry one apple and the information that there were originally 200. This is is data compression. In the end, you still have the information of 200 apples transferred, but you only had to carry a small amount with you during the transmission.

Lossless and lossy compression
A lossless compression would be able to accurately recreate the compressed digital signal. A lossy compression would remove so much data that it would be ultimately be irretrievable at the other end of the transmission. Lossy compression results in reduced picture quality in an entirely different way than when an analog signal degrades over generations, and in many ways it looks worse.

For most digital video systems, to be cost effective, compression is necessary because the computers don't have the processing power to handle the large amounts of data. Also, the high data rates overwhelm some computers and there can be rather unexpected crashing of the system. Of course, this is remedied by buying video-specific, high-end digital video equipment.

2. DIGITAL BASICS

Component and composite (analog)

Image origination sources, such as cameras and telecines, internally produce color pictures as three, full bandwidth signals—one each for Green, Blue and Red. Capitalizing on human vision not being as acute for color as it is for brightness level, television signals are generally transformed into luminance and color difference signals as shown in Figure 2-1. Y, the luminance signal, is derived from the GBR component colors, based on an equation such as $Y = 0.59\,G + 0.30\,R + 0.11\,B$. The color difference signals operate at reduced bandwidth, typically one-half of the luminance bandwidth. In some systems, specifically NTSC, the color difference signals have even lower and unequal bandwiths. Component signal formats and voltage levels are not standardized for 525 line systems, whereas there is an EBU document (EBU N-10) for 625 line systems. Details of component analog systems are fully described in another booklet from Tektronix: "Solving the Component Puzzle." It is important to note that signal level values in the color difference domain allow combinations of Y, B-Y, R-Y that will be out of legal gamut range when converted to GBR. Therefore, there is a need for gamut checking of color difference signals when making operational adjustments for either the analog or digital form of these signals.

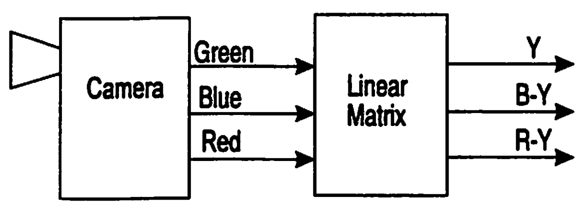


Figure 2-1. Component signals.

Further bandwidth reduction of the television signal occurs when it is encoded into composite PAL or NTSC as shown in Figure 2-2. Where each GBR signal may have a 6 MHz bandwidth, color difference signals would typically have Y at 6 MHz and each color difference signal at 3 MHz; however, a composite signal is one channel of 6 MHz or less. To carry this even further, the picture scanning is interlaced 2:1 to provide 60 pictures per second motion resolution with a 30 frames per second video rate. The net result is a composite 6 MHz channel carrying color pictures at a 60 per second rate that in its uncompressed form would have required three 12 MHz channels for a total bandwidth of 36 MHz. So, data compression is nothing new, digital just makes it easier.

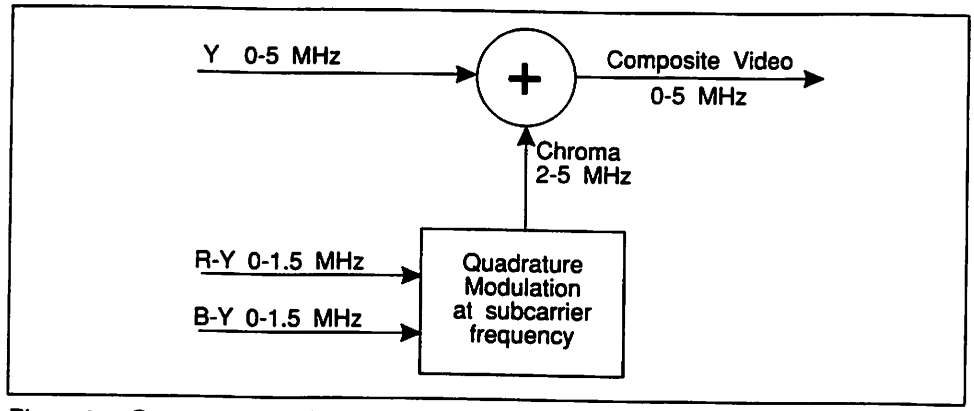


Figure 2-2. Composite encoding.

For composite NTSC signals there are additional gamut considerations when converting from the color difference domain. NTSC transmitters will not allow 100 percent color amplitude on signals with high luminance level (such as yellow). This is because the transmitter carrier will go to zero for signals greater than about 15 percent above 1 volt. Hence, there is a lower gamut limit for some color difference signals to be converted to NTSC for RF transmission.

As part of the encoding process sync and burst are added as shown in Figure 2-3. Burst provides a reference for decoding back to components. The phase of burst with respect to sync edge is called SCH phase, which must be carefully controlled in most studio applications. For NTSC, 7.5 IRE units of setup is added to the luminance signal. This poses some conversion difficulties, particularly when decoding back to component. The problem is that it is relatively easy to add setup, but removing it when the amplitudes and timing of setup in the composite domain are not well controlled can result in black level errors and/or signal disturbances at the ends of the active line.

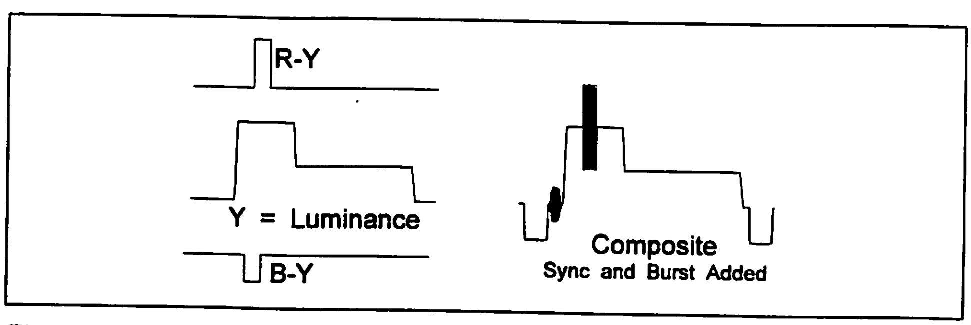


Figure 2-3. Color difference and composite waveforms.

Sampling & quantizing

The first step in the digitizing process is to "sample" the continuous variations of the analog signal as shown in Figure 2-4. By looking at the analog signal at discrete time intervals, a sequence of voltage samples can be stored, manipulated and, later, reconstructed.

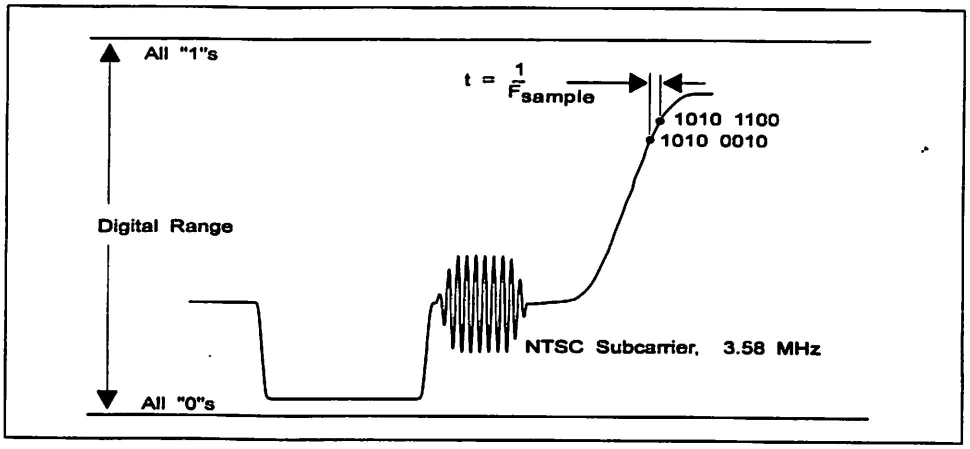


Figure 2-4. Sampling an analog signal.

In order to recover the analog signal accurately, the sample rate must be rapid enough to avoid missing important information. Generally, this requires the sampling frequency to be at least twice the highest analog frequency. In the real world, the frequency is a little higher than twice. (The Nyquist Sampling Theorem says that the interval between successive samples must be equal to or less than one-half of the period of the highest frequency present in the signal.)

The second step in digitizing video is to "quantize" by assigning a digital number to the voltage levels of the sampled analog signal—256 levels for 8-bit video, 1024 for 10-bit video, and up to several thousand for audio.

To get the full benefit of digital, 10-bit processing is required. While most current tape machines are 8-bit, SMPTE 125M calls out 10-bit as the interface standard. Processing at less than 10 bits may cause truncation and rounding artifacts particularly in electronically generated pictures. Visible defects will be revealed in the picture if the quantization levels are too coarse (too few levels). These defects can appear as "contouring" of the picture. However, the good news is that random noise and picture details present in most live video signals actually help conceal these contouring effects by adding a natural randomness to them. Sometimes the number of quantizing levels must be reduced; for example, when the output of a 10-bit processing device feeds an 8-bit recorder. In this case, contouring effects are minimized by deliberately adding a small amount of random noise (dither) to the signal. This technique is known as randomized rounding.

Digital video standards

Although early experiments with digital technology were based on sampling the composite (NTSC or PAL) signal, it was realized that for the highest quality operation, component processing was necessary. The first digital standards were component. Interest in composite digital was revived when Ampex and Sony announced a composite digital recording format, which became known as *D-2*. Initially, these machines were projected as analog in/out devices for use in existing analog NTSC and PAL environments; digital inputs and outputs were provided for machine to machine dubs. However, the post production community recognized that greater advantage could be taken of the multi-generation capability of these machines if they were used in an all digital environment.

CCIR 601

CCIR 601 is not a video interface standard, but a sampling standard. CCIR 601 evolved out of a joint SMPTE/EBU task force to determine the parameters for digital component video for the 525/59.94 and 625/50 television systems. This work culminated in a series of tests sponsored by SMPTE in 1981, and resulted in the well known CCIR Recommendation 601. This document specified the sampling mechanism to be used for both 525 and 625 line signals. It specified orthogonal sampling at 13.5 MHz for luminance, and 6.75 MHz for the two color difference signals C_B and C_R, which are scaled versions of the signals B-Y and R-Y.

The sampling structure defined is known as "4:2:2." This nomenclature is derived from the days when multiples of NTSC subcarrier were being considered for the sampling frequency. This approach was abandoned, but the use of "4" to represent the luminance sampling frequency was retained. The task force mentioned above examined luminance sampling frequencies from 12 MHz to 14.3 MHz. They selected 13.5 MHz as a compromise because the sub-multiple 2.25 MHz is a factor common to both 525 and 625 line system. Some extended definition TV systems use a higher resolution format called 8:4:4 which has twice the bandwidth of 4:2:2.

Parallel component digital

CCIR 601 described the sampling of the signal. Electrical interfaces for the data produced by this sampling were standardized separately by SMPTE and the EBU. The parallel interface for 525/59.94 was defined by SMPTE as SMPTE Standard 125M (a revision of the earlier RP-125), and for 625/50 by EBU Tech 3267 (a revision of the earlier EBU Tech 3246). Both of these were adopted by CCIR and are included in Recommendation 656.

The parallel interface uses eleven twisted pairs and 25-pin "D" connectors. (The early documents specified slide locks on the connectors; later revisions changed the retention mechanism to 4/40 screws.) This interface multiplexes

the data words in the sequence C_B, Y, C_R, Y, C_B . . . , resulting in a data rate of 27 Mwords/s. The timing sequences SAV and EAV were added to each line to represent Start of Active Video and End of Active Video. The digital active line contains 720 luminance samples, and includes space for representing analog blanking within the active line.

CCIR 601 specified eight bits of precision for the data words representing the video. At the time the standards were written, some participants suggested that this may not be adequate, and provision was made to expand the interface to 10-bit precision. Ten-bit operation has indeed proved beneficial in many circumstances, and the latest revisions of the interface standard provide for a 10-bit interface, even if only eight bits are used. Digital-to-analog conversion range is chosen to provide headroom above peak white and footroom below black as shown in Figure 2-5. Quantizing levels for black and white are selected so the 8-bit levels with two "0"s added will have the same values as the 10-bit levels. Values 000 - 003 and 3FF - 3FC are reserved for synchronizing purposes. Similar factors determine the quantizing values for color difference signals as shown in Figure 2-6.

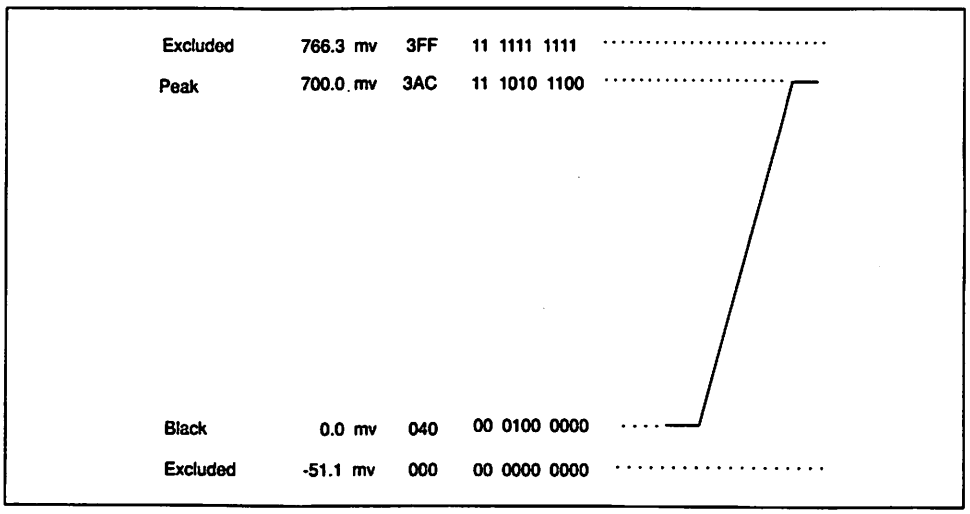


Figure 2-5. Luminance quantizing.

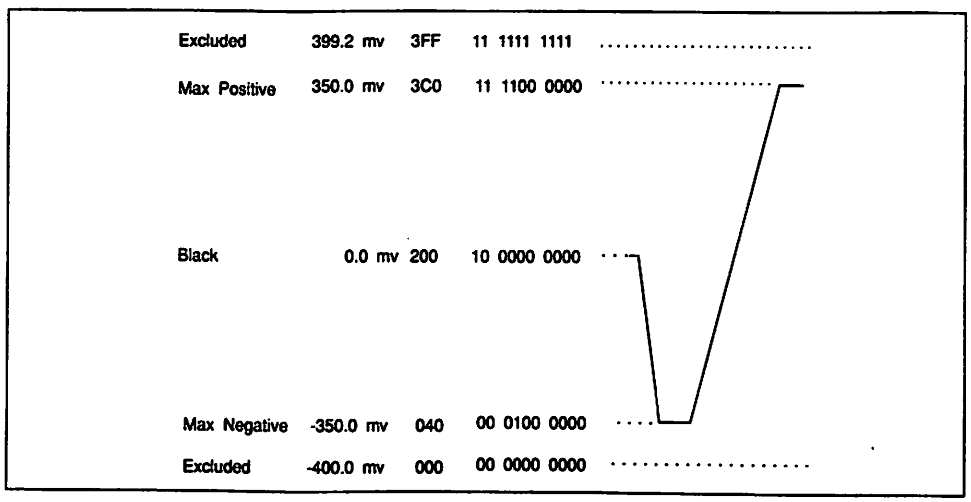


Figure 2-6. Color-difference quantizing.

Figure 2-7 shows the location of samples and digital words with respect to an analog horizontal line. Because the timing information is carried by EAV and SAV, there is no need for conventional synchronizing signals, and the horizontal intervals (and the active line periods during the vertical interval) may be used to carry ancillary data. The most obvious application for this data space is to carry digital audio, and documents are being prepared by SMPTE to standardize the format and distribution of the audio data packets.

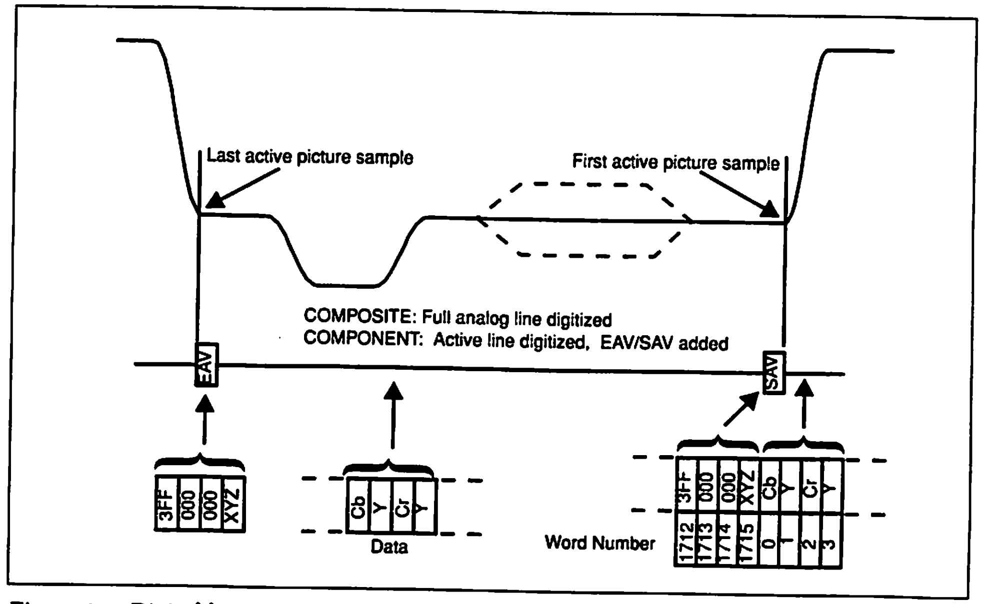


Figure 2-7. Digital horizontal line.

CCIR 601 is a well proven technology with a full range of equipment available for production and post production. Generally, the parallel interface has been superseded by a serial implementation, which is far more practical in larger installations. CCIR 601 provides all of the advantages of both digital and component operation. It is the system of choice for the highest possible quality in 525 or 625 line systems.

Parallel composite digital

The composite video signal is sampled at four times the (NTSC or PAL) subcarrier frequency, giving nominal sampling rates of 14.3 MHz for NTSC and 17.7 MHz for PAL. The interface is standardized for NTSC as SMPTE 244M and, at the time of writing, EBU documentation is in process for PAL. Both interfaces specify ten bit precision, although D-2 and *D-3* machines record only eight bits to tape. Quantizing of the NTSC signal (shown in Figure 2-8) is defined with a modest amount of headroom above 100% bars, a small footroom below sync tip and the same excluded values as for component.

PAL composite digital has been defined to minimize quantizing noise by using a maximum amount of the available digital range. As can be seen in Figure 2-9, the peak analog values actually exceed the digital dynamic range, which might appear to be a mistake. Because of the specified sampling axis,

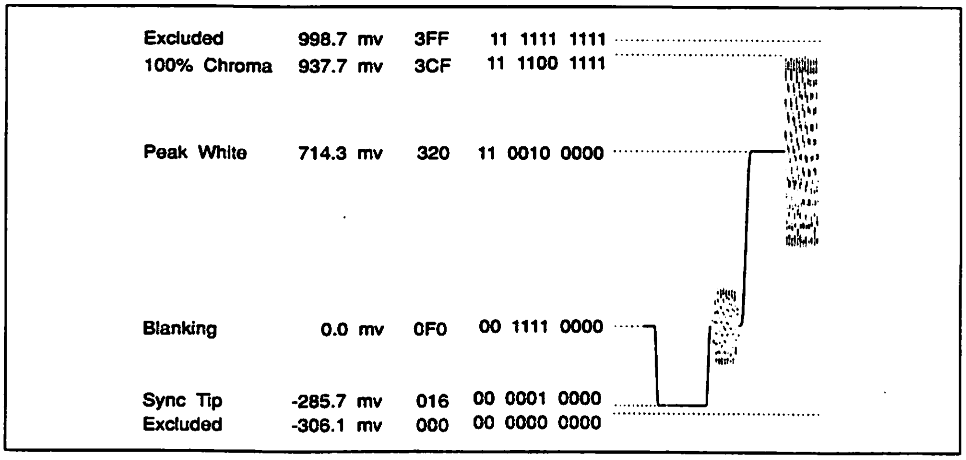


Figure 2-8. NTSC quantizing.

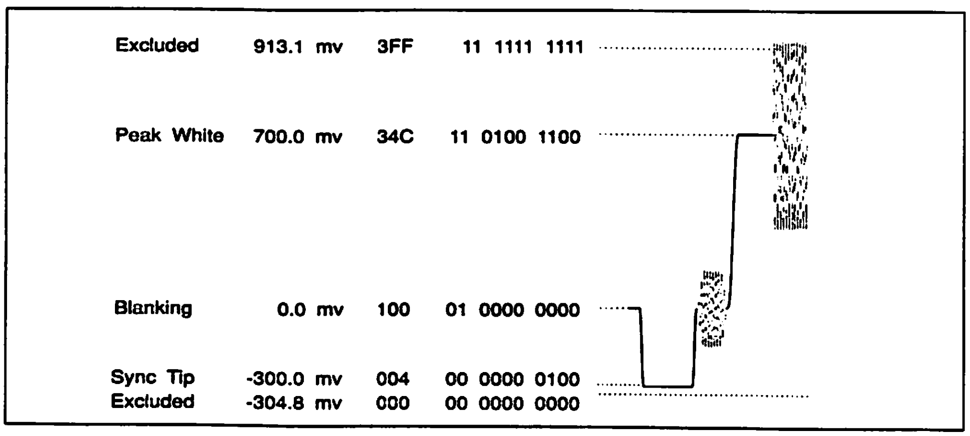


Figure 2-9. PAL quantizing.

reference to subcarrier and the phase of the highest luminance level bars (such as yellow), the samples never exceed the digital dynamic range. The values involved are shown in Figure 2-10.

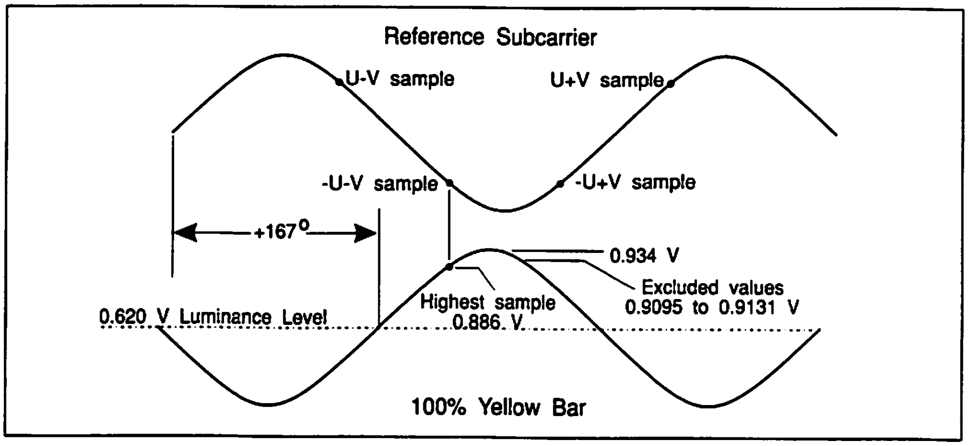


Figure 2-10. PAL yellow bar sampling.

Like the component interface, the composite digital active line is long enough to accommodate the analog active line and the analog blanking edges. Unlike the component interface, the composite interface transmits a digital representation of conventional sync and burst during the horizontal blanking interval. A digital representation of vertical sync and equalizing pulses is also transmitted over the composite interface.

Composite digital installations provide the advantages of digital processing and interface, and particularly the multi-generation capabilities of digital recording. However, there are some limitations. The signal is composite and does bear the footprint of NTSC or PAL encoding, including the narrow band color information inherent to these coding schemes. Processes such as chroma key are generally not satisfactory for high quality work, and separate provision must be made for a component derived keying signal. Some operations, such as digital effects, require that the signal be converted to component form for processing, and then re-encoded to composite. Also, the full excursion of the composite signal has to be represented by 256 levels on eight bit recorders. Nevertheless, composite digital provides a more powerful environment than analog for NTSC and PAL installations and is a very cost effective solution for many users.

As with component digital, the parallel composite interface uses a multi-pair cable and 25-pin "D" connectors. Again, this has proved to be satisfactory for small and medium sized installations, but practical implementation of a large system requires a serial interface.

Serial digital video

Parallel connection of digital equipment is practical only for relatively small installations, and there is a clear need for transmission over a single coaxial cable. This is not simple as the data rate is high, and if the signal were transmitted serially without modification, reliable recovery would be very difficult. The serial signal must be modified prior to transmission to ensure that there are sufficient edges for reliable clock recovery, to minimize the low frequency content of the transmitted signal, and to spread the transmitted energy spectrum so that radio frequency emission problems are minimized.

In the early 1980s, a serial interface for CCIR 601 signals was recommended by the EBU. This interface used 8/9 block coding and resulted in a bit rate of 243 Mb/s. Its interface did not support ten-bit precision signals, and there were some difficulties in producing reliable, cost effective, integrated circuits. The block coding based interface was abandoned and has been replaced by an interface with channel coding that utilizes scrambling and conversion to NRZI. The serial interface has been standardized as SMPTE 259M and EBU Tech. 3267, and is defined for both component and composite signals including embedded digital audio.

Conceptually the serial digital interface is much like a carrier system for studio applications. Baseband audio and video signals are digitized and combined on the serial digital "carrier" as shown in Figure 2-11. (It is not strictly a carrier system in that it is a baseband digital signal, not a signal modulated on a carrier.) The bit rate (carrier frequency) is determined by the clock rate of the digital data: 143 Mb/s for NTSC, 177 Mb/s for PAL and 270 Mb/s for CCIR 601 component digital. A widescreen (16x9) component system has been proposed that has a clock rate of 360 Mb/s.

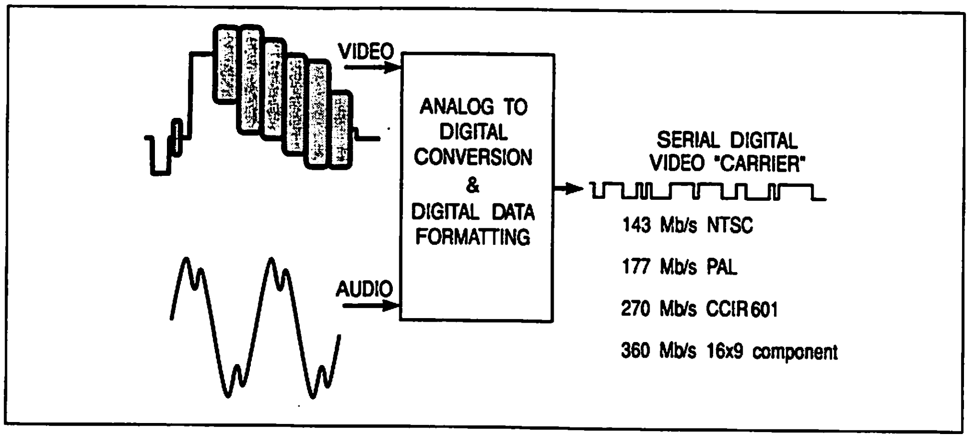


Figure 2-11. The carrier concept.

Parallel data representing the samples of the analog signal is processed as shown in Figure 2-12 to create the serial digital data stream. The parallel clock is used to load sample data into a shift register, and a times-ten multiple of the parallel clock shifts the bits out, LSB (least significant bit) first, for each 10-bit data word. If only 8 bits of data are available at the input, the serializer places zeros in the two LSBs to complete the 10-bit word. Component signals do not need further processing as the SAV and EAV signals on the parallel interface provide unique sequences that can be identified in the serial domain to permit word framing. If ancillary data such as audio has been inserted into the parallel signal, this data will be carried by the serial interface. The serial interface may be used with normal video coaxial cable.

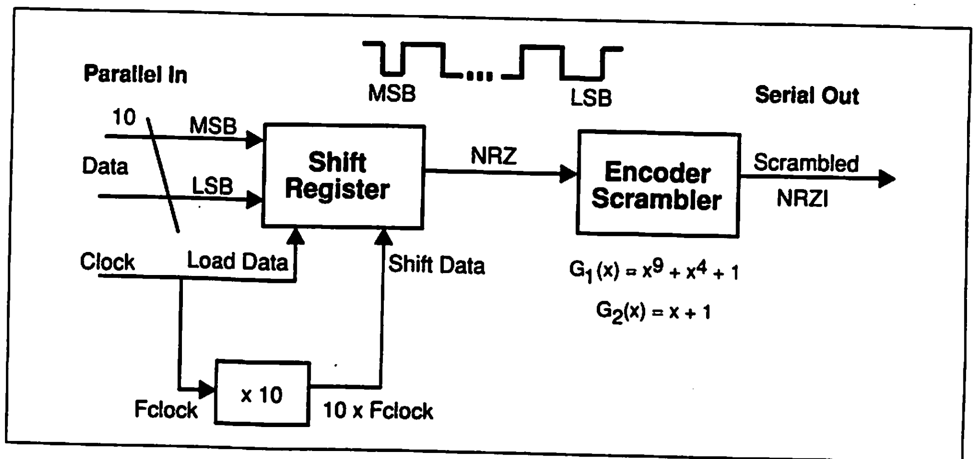


Figure. 2-12. Parallel-to-serial conversion.

The conversion from parallel to serial for composite signals is somewhat more complex. As mentioned above, the SAV and EAV signals on the parallel component interface provide unique sequences that can be identified in the serial domain. The parallel composite interface does not have such signals, so it is necessary to insert a suitable timing reference signal (TRS) into the parallel signal before serialization. A diagram of the serial digital NTSC

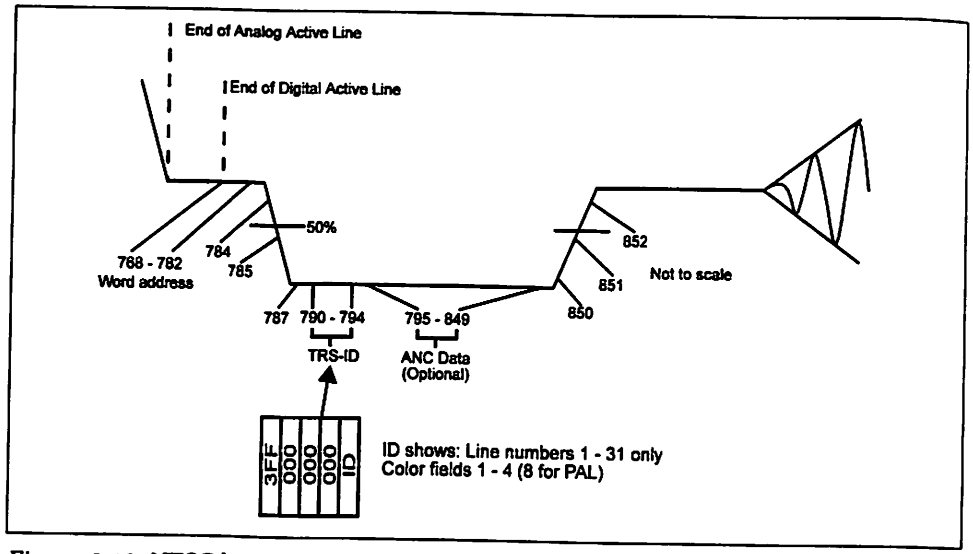


Figure. 2-13. NTSC horizontal interval.

horizontal interval is shown in Figure 2-13. The horizontal interval for serial digital PAL would be similar except that the sample location is slightly different on each line, building up two extra samples per field. A three word TRS is inserted in the sync tip to enable word framing at the serial receiver, which should also remove the TRS from the received serial signal.

The composite parallel interface does not provide for transmission of ancillary data, and the transmission of sync and burst means that less room is available for insertion of data. Upon conversion from parallel to serial, the sync tips may be used. However, the data space in NTSC is sufficient for four channels of AES/EBU digital audio. Ancillary data such as audio may be added prior to serialization, and this would normally be performed by the same co-processor that inserts the TRS.

Following the serialization of the parallel information the data stream is scrambled by a mathematical algorithm then encoded into NRZI (non-return to zero inverted) by a concatenation of the following two functions:

$$G_1(X) = X^9 + X^4 + 1 \qquad G_2(X) = X + 1$$

At the receiver the inverse of this algorithm is used in the deserializer to recover the correct data. In the serial digital transmission system the clock is contained in the data as opposed to the parallel system where there is a separate clock line. By scrambling the data, an abundance of transitions is assured as required for clock recovery. The mathematics of scrambling and descrambling lead to some specialized test signals for serial digital systems that are discussed later in this booklet.

Encoding into NRZI makes the serial data stream polarity insensitive. NRZ (non return to zero, an old digital data tape term) is the familiar circuit board logic level, high is a "1" and low is a "0." For a transmission system it is convenient to not require a certain polarity of the signal at the receiver. As shown in Figure 2-14, a data transition is used to represent each "1" and there is no transition for a data "0." The result is that it is only necessary to detect transitions; this means either polarity of the signal may be used. Another result of NRZI encoding is that a signal of all "1"s now produces a transition every clock interval and results in a square wave at one-half the clock frequency. However, "0"s produce no transition, which leads to the need for scrambling.

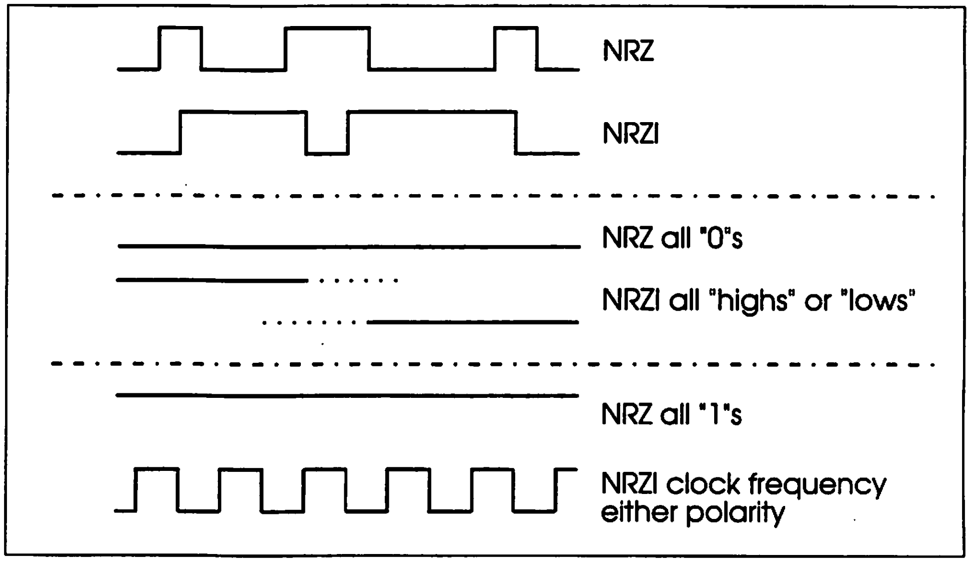


Figure. 2-14. NRZ and NRZI relationship.

Rate conversion - Format conversion

When moving between component digital and composite digital in either direction, there are two steps: the actual encoding or decoding, and the conversion of the sample rate from one standard to the other. The digital sample rates for these two formats are different: 13.5 MHz for component digital and 14.3 MHz for NTSC composite digital (17.7 MHz for PAL). This second step is called "rate conversion." Often the term rate conversion is used to mean both encoding/decoding and resampling of digital rates. Strictly speaking, rate conversion is taking one sample rate and making another sample rate out of it. For our purposes we'll use the term "format conversion" to mean both the encode/decode step and resampling of digital rates. The format conversion sequence depends on direction. For component to composite, the usual sequence is rate conversion followed by encoding. For composite to component, the sequence is decoding followed by rate conversion. See Figure 2-15.

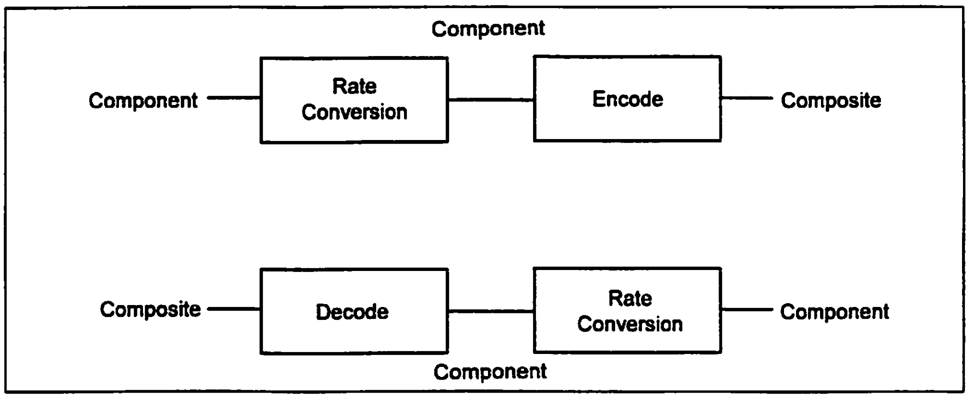


Figure. 2-15. Format conversion.

It is much easier to do production in component because it isn't necessary to wait for every color framing boundary (four fields for NTSC and eight for PAL) to match two pieces of video together; instead, they can be matched every two fields (motion frame). This provides four times the opportunity. Additionally, component is a higher quality format since luminance and chrominance are handled separately. To the extent possible origination for component environment production should be accomplished component. A high quality composite-to-component format converter provides a reasonable alternative.

After post-production work, component digital often needs to be converted to composite digital. Sources that are component digital may be converted for input to a composite digital switcher or effects machine. Or a source from a component digital telecine may converted for input to a composite digital suite. Furthermore, tapes produced in component digital may need to be distributed or archived in composite digital.

Computers

Computers, like telephones and televisions, are quickly populating businesses and households everywhere. However, it should be noted that computers are also discretely used in automobiles, calculators, VCR's, airplanes, grocery stores, etc. This implies the multipurpose nature of computers. They can be used for just about anything. But what are they?

computer - "an electronic apparatus that can receive, process, store, and retrieve data, can carry out mathematical and logical operations at high speed and display the results and be programmed"

Computers are devices that can do mathematical operations at high speeds. They are the ultimate manipulators and processors of information. And they are programmable. In other words, a computer is not built for a specific function like a screwdriver is. It is like a number of processing blocks which can be arranged by the user for a specific purpose. Really, a computer only does one thing: mathematical operations as determined by the user. What distinguishes one computer from another is the allowable size of the numbers to be processed and the speed at which operations are performed on the numbers.

Speed and number crunching

Imagine sitting in elementary school math class. The teacher asks you to add ten two-digit numbers together. It seems like a bit of a challenge but you can manage it. But then things get out of hand: the teacher requests that you calculate the total of one thousand two-digit numbers together. As soon as you finish, another request is made to calculate a thousand *ten-digit* numbers! "What do you think I am, a computer?" you say.

Computers perform one primary function: operate on numbers over time. A modern computer can do one million instructions per second. High-end computers can do a billion instructions per second if not more. A calculation that once took days on paper would suddenly be reduced to a fraction of a second with a computer. Which numbers are operated on and how are determined by the computer programmer, who designates the order of events.

Computer decision-making, logic

Computers are capable of making decisions based on the rules of logic. In logic, all decisions are reduced to variables being either true or false. This black-or-white thinking makes the fundamental nature of computers very simple. Every choice is like a fork in the road: you can go one way or the other, but never in between. The path of decision-making is thus logical and easy to follow.

Numerically, true is given the value of 1 and false is given the value of 0. These two values digitally (numerically) represent logic.

Representation

Computers are multifaceted, and yet their only means of representation is low-high, off-on, false-true, 0-1. Any information can be represented in a computer but it must be done with either a true or false (0 or 1).

People numbers and computer numbers

Human beings have generally accepted a certain type of numerical system. It is called the *decimal system* and it is based on ten values... 0, 1, 2, 3, 4, 5, 6, 7, 8, 9. There are, however, other numerical systems.

System	Numerical base	System symbols
Binary	base 2	0, 1
Octal	base 8	0,1, 2, 3, 4, 5, 6, 7
Decimal	base 10	0, 1, 2, 3, 4, 5, 6, 7, 8, 9
Hexadecimal	base 16	0, 1, 2, 3, 4, 5, 6, 7, 8, 9, A, B, C, D, E, F

The decimal system is so widely accepted that we forget that it is just a way of coding *how much*, how many, or the amount of something. This is how the decimal code works:

If there are 157 french fries in a bag, there are really (1 x 100) + (5 x 10) + (7 x 1) = 157. Notice that there is a one's place, a ten's place, a hundred's place, and so on. Each place is ten times larger than the one to the right. This is how amounts are coded into a base-10, decimal system.

What about a different system? The hexadecimal system is identical except that each place is 16 times larger than the one to the right. There is a one's place, a sixteen's place, a 256's place, etc. Here are some decimal to hexadecimal equivalents:

Decimal	Hexadecimal	
5	5	5 (x 1) = 5
10	A	A (x 1) = 10
15	F	F (x 1) = 15
16	10	1 (X 16) + 0 (X 1) = 16
100	64	6 (X 16)+ 4 (X 1) = 100
200	C8	C (X 16) + 8 (X 1) = 200
251	FB	F (X 16) + B (X 1) = 251
300	12C	1 (X256) + 2 (X16) + C (X1) = 300

(A = 10, B = 11, C = 12, D = 13, E = 14, F = 15)

The binary system
Binary counting is based on two. Therefore, there are only two numerical symbols, 0 and 1. There is a one's place, a two's place, a four's place, and eight's place, a sixteen's place and so on. Each place is two times as large as the one to the right.

Decimal	Binary	
1	0001	0 (x8)+0 (x4)+0 (x2)+1 (x1) = 1
2	0010	0 (x8)+0 (x4)+1 (x2)+0 (x1) = 2
3	0011	0 (x8)+0 (x4)+1 (x2)+1 (x1) = 3
4	0100	0 (x8)+1 (x4)+0 (x2)+0 (x1) = 4
5	0101	0 (x8)+1 (x4)+0 (x2)+1 (x1) = 5
6	0110	0 (x8)+1 (x4)+1 (x2)+1 (x1) = 6
7	0111	0 (x8)+1 (x4)+1 (x2)+1 (x1) = 7
8	1000	1 (x8)+0 (x4)+0 (x2)+0 (x1) = 8
9	1001	1 (x8)+0 (x4)+0 (x2)+1 (x1) = 9
10	1010	1 (x8)+0 (x4)+1 (x2)+0 (x1) = 10
15	1111	1 (x8)+1 (x4)+1 (x2)+1 (x1) = 15

Representing 0's and 1's in real life
There are a number of ways of representing 0 and 1, off and on. An easy example is alight switch. When the switch is turned off, the light bulb is off. When on, electricity flows through the circuit and the light bulb is on. 0's and 1's can be represented by turning a switch on and off, sending electricity of a high voltage or almost no voltage. In real life computer systems, +5 volts often represents a 1 and 0 volts represents a 0.

Voltage can represent 0's and 1's and thus electricity can be used to represent different values. Based on the binary value table above, a decimal value of 5 can be represented to a computer electrically by a low voltage pulse follow by a high voltage pulse followed by a low voltage pulse and finally another high voltage pulse. In other words, low-high-low-high is a DIGITAL ELECTRICAL CODE for the decimal value 5.

Bits, bytes, words
A bit is the smallest amount of binary information. It can have one of two states: on or off. According to the table above, it takes four bits to represent decimal values of 8 or higher. The more bits available, the more places are available to represent higher numbers. 1 bit codes the one's place. 2 bits codes the two's place and the one's place. A third bit codes the four's place. A fourth bit represents the eight's place, and so on.

A byte is eight bits. This means a byte can represent 256 decimal values. Two bytes can represent 65,536 decimal values. Three bytes (or 24 bits) can represent any of 16,777,216 decimal values. Each bit added doubles the highest possible value of a binary system. Each byte added increases the highest possible value by a factor of 256. It is important to see the difference between a decimal representation and a binary representation:

16,777,215 (decimal, 8 places) = 111111111111111111111111 (binary, 24 places)

A binary word is a pre-defined number of bits which represents information. A word can be 10 bits, 12 bits, 20 bits, 32 bits, etc. It is dependent on the information represented. A ten bit word, for example, might represent brightness values of one point of light in time on a screen. The brightness values would range between 0 and 1023, or 000000000 and 1111111111.

Another type of word might not represent numerical data but be a code, like a ten bit synchronization word. A digital synchronization pulse might simply be a ten bit word spelled out 1111100000. Every time a computer read a ten bit 1111100000, it would translate this to mean a synchronization pulse and act accordingly.

Decisions and math

The nature of 0's and 1's is that they can mean literally anything... decimal numbers, communication codes like synchronization, true or false. A computer can be used for any application where quick mathematical processing is needed and automatic decisions can be made based on the data input.

All a computer requires is that the input information is in a digital, 0's and 1's format. Electrically, this means sending a consistent high voltage to represent 1 and a low voltage to represent 0.

Parts of a Computer

Over the last forty years, a standard computer system design has been developed.

CPU - the central processing unit. This is the main computer, or brains of the system. It is the part capable of doing thousands, millions, or billions of instructions per second. It takes binary data in and operates on it based on the programmed instructions.

ROM - read-only memory. ROM is the untouchable storehouse for the computers basic operating procedures, or source code. In other words, ROM contains the basic instructions which the CPU uses to process input. When the computer boots up, it looks for it's first instructions in the ROM.

RAM - random access memory. RAM is a large grid of data storage which the CPU can access at any time. This is the most quickly accessible data storage for the CPU, so the more RAM in the computer, the faster things can be processed. RAM is made up a high-density Integrated Circuits (IC's) which are really thousands of tiny transistors packed into a very tiny area. RAM is expensive because the high density expectation produces a large failure rate in the factory.

Input - a means of getting data into the computer. A keyboard, mouse, and disk drive are means of getting data into a digital form and into the computer for processing.

Output - computer monitor, disk drive, printer. This is where the digital data is translated back into usable form as words or numbers on a screen or paper.

Busses - connections between the computer components. The wider the bus, the more information will be able to travel at a given time. A one-bit bus would only allow one 0 or 1 to travel at any one time. A thirty-two bit bus would allow a data transfer of thirty-two bits from RAM to the CPU at one moment. A one-bit bus would take thirty-two times as long. Busses can be compared to highways: the number of lanes is the bit-width (how many bits may travel at one moment) and the bits themselves are the traffic, known as the bitstream.

CPU

In general terms, the CPU is considered to be the actual box which houses all of the main components: RAM, ROM, internal drives, specialized circuit boards for video or audio processing, and input/output connectors.

Memory

When a computer is first turned on, it needs to know what logical operations to do first. Without any start-up - or "booting" - instructions, flipping the on switch would give the computer power but not give it access to how to initialize itself. The basic instructions that cause the predictable nature of a computer start-up must be stored somewhere. Just as you know where the kitchen is as soon as you wake up in the morning, a computer knows where it's basic instruction set is. It is stored in a Read Only Memory which does not disappear even when the power turns off. This memory is unalterable, thus the name Read Only. It is called ROM.

ROM is where the basic instruction sets of the computer are located.

The "system" of a computer

A computer system is the virtual platform that a user interacts with. It may be a simple ">" prompt awaiting commands from the user, or a very easy-to-use desktop presentation which allows opening and closing of files and folders by pointing and clicking a mouse.

The basic system is NOT in a computer's ROM. It must be read from a source such as a hard drive, CD-ROM, or floppy drive.

Floppy drives, hard drives, compact discs, magnetic tape

All computer data storage devices have one thing in common: they store 0's and 1's. They are distinguished by their writing and retrieval speed of data and by the method of storage.

The fastest reading/writing computer storage devices is a hard metal disk spinning in an enclosed shell. It is somewhat like a magnetic record player, where a needle can access any point on the disk at any time. This makes data "randomly accessible", meaning that at any random point of time, a specific point can be accessed on the disk. Hard disks are random access devices.

Floppy disks are similar in principle but they are encased in a plastic case and are flexible as opposed to rigid.

Compact discs use optical technology (lasers) to read and write O's and 1's onto a rigid plastic disc. Though it is a random access medium, it is much slower than a hard disk.

Magnetic tape, used in audio and video production, is also used to store digital data. Tape is definitely not a random access medium, but it can store large amounts of data because of it's relatively large dimensions compared to a hard disk.

Monitor
Computer monitors work much the same way a TV monitor works. A main difference is that the scanning is not interlaced (odd lines, then even lines scanned). The scanning is also at a different rate than broadcast TV. Interestingly, a computer sends Red, Green, Blue and sync (horizontal and vertical) to a monitor instead of color difference signals, Y/C, or NTSC. This means a computer video OUTPUT is not readily compatible with a composite NTSC signal. You could not attach a VCR to the monitor port and expect to get a usable signal.

The screen resolution is 72dpi (dots per inch).
The number of colors on your screen will depend on the bit-depth, or how many bits represent each pixel. The more bits per pixel, the more subtle the gradations of brightness, hue and chrominance.

1-bit displays: black and white, each pixel is on or off
2-bit displays: black, 2 shades of gray, white
8-bit displays: 256 levels of gray OR 256 colors in a color palette.
24-bit displays: over 16 million color possibilities

The size of the screen and the bit-depth will determine the amount of Video RAM (VRAM) necessary for the computer. A large screen takes more memory to keep all the pixels of the screen updated, and more bit-depth requires even more yet.

Speakers
Modern computers tend to have external speakers driven by an audio output port. Speakers often have volume controls.

Input / Output ports
The printer port is a serial port (one bit at a time) for connecting printers or making a connection to an Appletalk network (Macintosh).
The modem port is a serial port for connections to an external modem.

The SCSI (Small Computer Systems Interface) is a wide, multi-wire connection used to attach external hard-drives, CD-ROM players and burners, Syquests, DataDAT drives, scanners, etc.
SCSI devices may be daisy-chained together, but special rules must be followed. Each device gets a unique SCSI ID number from 1 to 6 (0 being reserved for the CPU and 7 for the computer's internal hard drive). Also, the first and last item in the chain must have a proper electrical resistance to create an acceptable circuit. This is known as termination. The first device in the chain is typically the computer's internal hard drive and it is already terminated. The last device will need a terminator unless it is actively terminated, which means you simply flip a switch to turn on the terminator. The most advanced SCSI devices are self-terminating depending on what is connected to it.

The monitor port will drive one computer monitor (size dependent on amount of memory available on particular computer) by sending Red, Green, Blue and Sync signals.

Ethernet port is for connecting computers into a Local Area Network so that files and programs may be shared between "desktops". This means files such as audio, video, still images and text files can be traded and worked on simultaneously at several workstations.

Audio ports: there are usually mini or RCA audio jacks on modern personal computers. However, the analog-to-digital converters are usually mediocre. It is better to buy an specific purpose external card to install into the computer, such as one for audio or video.

Video ports: some computers have RCA connector video ports that allow NTSC video in or out. Some also have S-Video (Y/C) connections.

Space for more: there are often empty slots on the computer motherboard and open spaces at the back of the computer where a card can be installed whose connectors will come out the back of the computer. This could be an audio, video, second monitor driver, second SCSI connector, etc.

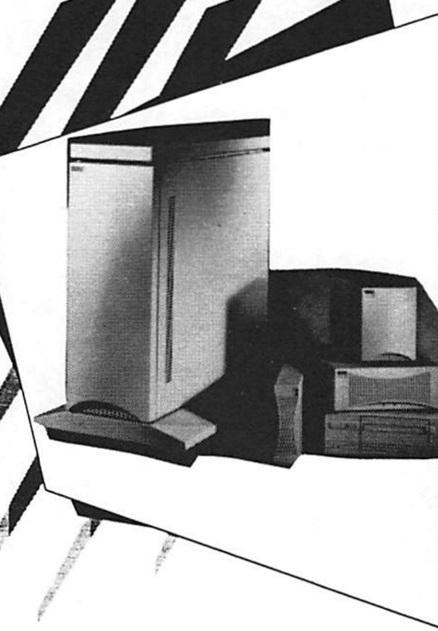
Additional hardware
Often, additional hardware can be bought and installed into a computer to increase its functionality and to make it more of a specialized device. Cards are installed into empty slots on the computers main "motherboard" and any connectors that may be on it will come out the back of the computer. Avid uses several internal cards: one for digitizing video (analog-to-digital converter), one for compression (no connectors; connects directly to A-to-D converter), and a SCSI accelerator card (adds a new SCSI connector to back of Macintosh which is faster than the built in one).

Faster SCSI: this is useful when dealing with high sampling rates, for example digital audio. Even more intensive is digital video. The SCSI connection must be fast enough to write all of the digital information onto a hard drive.

ProTools is another example of additional hardware added to make a specific device. There is an analog-to-digital / digital-to-analog converter which attaches to the motherboard. A rackmountable unit, called a breakout box, connects to the card for more elegant access to connectors. There is also a digital signal processor card (DSP) to do audio effects work, mixing and equalization... all digitally within the computer. There is also a SCSI accelerator card for rapid "throughput" of digital data from sampling to writing to playback off the hard drive.

AVIDdrive™

Storage Capacities for Avid's Film, Video and Audio Post-Production Products



Single-Field Resolutions

Frames per Second	Audio KHz	Audio Channels	MINUTES PER GB AVR1 Range	AVR2 Range	AVR3 Range	AVR4 Range	AVR5 Range
30/25	0	0	63-107	44-85	30-56	24-48	14-22
30/25	11	1	58-94	41-76	29-52	23-46	14-22
30/25	11	2	54-84	39-69	28-49	22-43	13-21
30/25	44.1	1	47-69	36-59	26-43	21-39	13-20
30/25	44.1	2	38-51	30-45	23-35	19-32	12-18
30/25	48	2	37-48	29-43	22-34	19-31	12-18
24	0	0	79-134	54-106	37-69	30-61	17-28
24	11	2	65-99	48-83	34-59	27-52	17-26
24	44.1	1	56-79	42-68	31-51	26-46	16-24
24	48	1	54-76	42-66	31-50	25-45	16-24

Enhanced, Single-Field Resolutions

Frames per Second	Audio KHz	Audio Channels	MINUTES PER GB AVR1e Range	AVR2e Range	AVR3e Range	AVR4e Range	AVR5e Range	AVR6e Range
30/25	0	0	71-116	47-90	40-69	26-54	14-31	9-17
30/25	11	1	65-101	44-81	38-63	25-51	14-30	9-17
30/25	11	2	60-89	42-73	37-59	24-48	14-29	9-17
30/25	44.1	1	52-72	38-61	33-51	23-42	13-27	8-16
30/25	44.1	2	41-53	32-47	29-40	20-35	12-24	8-15
30/25	48	2	39-50	31-45	28-39	20-34	12-23	8-14
24	0	0	89-145	59-112	51-86	32-68	18-39	11-22
24	11	2	72-105	51-87	45-71	30-58	17-36	11-20
24	44.1	1	61-83	45-71	40-60	28-50	16-33	10-19
24	48	1	59-80	44-69	39-58	27-49	16-33	10-19

NOTES:

- Minutes per GB are approximately the same for NTSC (30 fps) and PAL (25 fps).
- Highest number in range represents simple footage such as talking heads, medium angle shot or simple backgrounds.
- Lowest number in range represents complex footage such as a group of people, outdoor motion or complex backgrounds.
- Letterbox at 24 fps will give you more minutes of footage per GB.
- Minutes per GB are based on Media Composer® 5.2 software.
- Ranges for 24 fps apply only to NTSC format.

Two-Field Resolutions

Frames per Second	Audio KHz	Audio Channels	MINUTES PER GB AVR25 Range	AVR26 Range	AVR27 Range
30/25	0	0	9-19	6-9	4-9
30/25	11	1	9-18	6-9	4-9
30/25	11	2	9-18	6-9	4-9
30/25	44.1	1	8-17	5-8	4-8
30/25	44.1	2	8-16	5-8	4-8
30/25	48	2	8-15	5-8	4-8

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FOREWORD

Digital television has been intensively studied by broadcasters and industry alike since the early 1960's. Very early on it was recognised that the need to record digital video and audio signals was the most significant factor necessary to ensure the widespread use of digital processing for professional television applications.

During the development of digital encoding standards for television signals it was possible, and indeed necessary, to consider in-depth the conflicting requirements for a digital standard in that it must provide an adequate subjective picture quality and yet enable the data to be practicably and economically recorded. This is in contrast to the earlier analogue situation where there was no opportunity to achieve an optimum solution for the video recorder because it had to be designed to cope with an already existing television signal.

On the other hand it is clear that, by its very nature, digital processing demands a more disciplined approach to system design needing many detailed parameters to be specified before equipment design is begun. One needs only to refer to the computer industry to recognise that many systems have been difficult to implement because of poor data compatibility and lack of a common interface between equipments.

Hence the development of the digital tape recorder has been carried out more or less in parallel with that for the encoding parameters and interface specifications for the digital video and the digital audio. Of most importance is that the ensuing standards have received wide international acceptance and ratification by the CCIR, the body responsible for the consolidation of broadcast engineering recommendations, to enable international programme exchange.

A unique feature of this work has been the collaboration between experts from the broadcast industry (the users) and manufacturing industry. The main standardizing bodies have been the European Broadcasting Union (EBU) and the Society of Motion Picture and Television Engineers (SMPTE) in the USA. In both cases these organizations formed specialist committees known, for the EBU by the acronym MAGNUM (MAGnetoscope NUMérique – the French term for a digital magnetic recorder), and for the SMPTE equivalent as the DTTR (Digital Television Tape Recording) Working Party. Also worthy of note is the remarkably close liaison between these two committees resulting in a joint approach to a unified standard that could be put to the 1986 CCIR Plenary Session.

During the development a major question was how radical should the proposals be? We are well aware how rapid progress in technology can make

what appears to be extremely difficult to be almost trivial subsequently. Thus, one could search for an ambitious result to avoid early obsolescence, but equally many engineering failures have occurred simply because the technology was ahead of its time. In the generation of the standard for the digital video tape recorder great care was taken to see that forward projections were ambitious but realistic and not based on the principle that something might turn up.

On the other hand simple arithmetic demonstrated very much higher tape packing densities were needed than had ever been envisaged for analogue video tape recorders. To begin with, the total analogue bandwidth specified in CCIR Recommendation 601 for the component signals (comprising a luminance and two colour difference signals) is 13.5 MHz. This is approximately 2.5 times that needed for the PAL I composite signal. Additionally the user opinion expressed in the committees increasingly favoured a cassette recorder rather than an open-reel version. Maintaining equivalent tape packing densities to the one inch type C machines, would result in truly enormous cassettes, even if one ignores the extra overhead in the conversion to the digital form. It was known that increased density could be obtained, without loss of signal-to-noise ratio, by using narrower and, consequently, longer tracks. This in turn demands increased tracking accuracy. Reducing the tape thickness was also seen to be a means by which the size of cassettes may be reduced, but thinner tape requires more careful handling in the tape transport. These and many other interacting factors were central issues in the creation of the standard.

It is unfortunately traditional to assume that audio is treated almost as an afterthought in the design of analogue video tape recorders. Not so in the digital video tape recorder. It was quickly realised that audio quality which could stand comparison with that of the digital video was not likely to be achieved using conventional longitudinal tracks, at least in the foreseeable future. Digital techniques, however make it relatively simple to contemplate multiplexing the digital audio in with the digital video. For editing purposes the video and audio need to be recorded in separate bursts rather than interleaved together, but without losing the attractive feature of being able to use the same record and replay heads for both signals. Comparison of the relative sizes of the video and audio bursts to those of possible scratches and other tape damage led to the conclusion that the audio data should be given the most secure position on the tape. This resulted in the idea (at first sight rather strange) of recording the audio bursts in the centre of the tape.

The realisation of a digital video recorder is, therefore, so very different from its analogue counterpart, in spite of a superficially similar physical resemblance, that it is certain that a great deal of background information must be supplied to those who will experience the new recorders in their professional life. This book is provided as a contribution to the bridging of the gap between the informed expert and interested user and comes from people who have been closely involved in digital video tape recording in its formative years.

In concluding this Foreword, I must mention the names of both Aleksandar

CHAPTER 1

4:2:2 DVTR – REQUIREMENTS AND FORMAT OVERVIEW

*by Aleksandar Todorović**

1.1 TOWARDS THE ALL-DIGITAL STUDIO

Over the past 40 years or so most of the technological advances in electronics have been made in the analogue domain. Recent work has, however, concentrated on improving digital technology especially in the fields of memory and processing circuitry. The advances in digital devices and techniques, developed mainly for the computer industry, have been introduced into television to bring major advances in digital control and processing. The use of digital signal processing in the broadcast television industry has been recognized as the opening of a new era.

It was clear from the outset that the digital approach would permit the creation of new and considerably more versatile production tools. The possibilities for creative programme making would be broadened and the quality of sound and picture would be dramatically improved.

To exploit this new technology to its maximum it is necessary to create and operate a totally digital studio, where the video signal is digitized immediately after the electro-optical conversion and kept in that form throughout the whole programme-making process. The final programme should be stored in digital form, and the digital-to-analogue conversion carried out at the input to the distribution network, or even at the input to the transmitter. In order to achieve this goal, two basic problems need to be solved:

(1) The precise definition of the nature of video and audio signals in their digital form, i.e. to define studio interface signals.
(2) The development of the necessary studio equipment.

It has taken almost 10 years to solve the first of these two problems. The correct definition of the goal and the attempts to reach it went in parallel with a rapid development of the technology. From the initial modest task of converting the composite colour video signal into a digital form in such a way as to obtain a sufficiently low bit-rate (to enable economical trans-

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mission over international microwave links), the goal moved to the idea of an all-digital television studio. At this point it soon became clear that important advantages could be obtained by abandoning the use of a composite colour video signal and using instead colour components. This was a crucial move which will have further repercussions even for analogue television.

The possibility of defining a new studio interface standard, based on colour components, was perceived as a unique opportunity to bring differing world television standards closer together. It should be realised, however, that the production of television programmes covers a very broad spectrum, and quality standards, which may be required or accepted, may vary greatly. At one end of the spectrum there is the news gathering activity, where it is possible to compromise on quality because of the immediacy of the programme content and to permit the use of lightweight and portable equipment. At the opposite end there is the elaborate studio production and post-production facilities which may require equal bandwidth component signals to allow all manner of signal manipulation while at the same time maintaining the highest possible signal quality.

The result of all these considerations was the establishment of the family concept. This concept provides for three levels each with its own parameters. These parameters are, however, selected in such a way as to have a simple arithmetical relationship, which facilitates easy transcoding from one level to another. The central, or studio, level is also known as the 4:2:2 level since there is a two-to-one relationship between the sampling frequencies for the luminance and the two colour difference signals. The lower level (still undefined in the international standard in 1986) may be derived by halving all the sampling frequencies, i.e. 2:1:1. At the higher level of the family (also still undefined) all three components have equal bandwidths and consequently it is known as the 4:4:4 level, i.e. the sampling frequencies for all three components are identical.

As mentioned previously the aim was to bring different television standards closer together. In reality it was necessary to define digital coding parameters for each of the two existing television standards, i.e. 625/50 and 525/60. The only possible way to marry these contradictory requirements was to define the parameters in such a way as to ensure they were as similar as possible. The 4:2:2, or studio level, defines, for both 625/50 and 525/60 stadards, a common sampling frequency of 13.5 MHz for the luminance signal and 6.75 MHz for each of the colour difference signals, together with a common number of active samples per line (720 for the luminance signal and 360 for each of the two colour difference signals), a common number of bits per sample and the same number of quantization levels. All these parameters are now part of CCIR Recommendation 601.

This definition of the digital interface standard required a source bit rate of 216 Mbit/s. To build television studio equipment capable of handling this bit-rate is not an easy task. It is true that electronic technology is progressing at a very fast rate (it took fewer than six years to pass from a 1K RAM chip to a 64K RAM chip) but, memorizing, handling and processing at such high

bit-rates nevertheless requires the use of the most advanced technology of the day and even some emerging technology of tomorrow.

Since its appearance in 1956 the video tape recorder (VTR) has acquired the key role in any production studio. The VTR is necessary and essential in recording, post-processing and transmission of television programmes. It is quite clear that an all-digital television studio simply cannot exist without a digital video tape recorder (DVTR). The DVTR is the corner stone of the new edifice. It was for this reason that industry and broadcasters put such a great effort into the definition of a digital video tape recording standard. After six years of intensive work the goal was reached in 1986 when the CCIR adopted the proposed digital recording format as a new Recommendation.

1.2 LIMITATIONS OF ANALOGUE RECORDING

One of the major advantages of moving from the analogue to the digital domain is the possibility of overcoming some of the basic limitations imposed by analogue techniques. This is particularly true of magnetic video tape recording where the quality of the final copy is to a large extent defined by the limitations of the analogue process.

Signal-to-noise ratio rapidly decreases as the number of analogue generations increases. Furthermore, different parameters, such as differential gain-and-phase, chrominance/luminance delay, the head-band edge-effect and velocity errors, have a cumulative (and deteriorating) effect in successive generations. The multi-generation capability of even the best professional analogue recorders (under ideal operating conditions) is limited to a modest five to six generations.

Sound quality, generally treated as a poor relation in television, receives even worse treatment in analogue video tape recorders where its quality lags far behind that possible with analogue audio tape recorders. In the multi-generation process the degradation of audio-quality is at least as fast as the degradation of video quality.

Analogue composite video tape recording imposes an additional limitation. In the editing process it is necessary to maintain the PAL eight-field sequence which thus reduces the editing definition to four pictures. The problem also arises with the NTSC system which has a four-field sequence and hence an editing definition of two pictures.

Finally, the analogue composite output of present day studio recorders permits only a very limited range of signal manipulation, which today is the essence of any elaborate post-production.

1.3 DIGITAL VIDEO TAPE RECORDING

Digital video tape recording opens up the possibility of overcoming the limitations of analogue recording and providing features unachievable in the

analogue domain. The quasi-transparency of the digital recorder dramatically improves the number of high quality generations possible, the digital recording of sound signals leads to the highest possible audio quality, the component approach overcomes the problem of the PAL/NTSC sequence and the reproduced digital component signal can be manipulated in every possible way, from down-stream chroma-keying to colour correcting and picture re-framing.

At the same time, the digital video tape recorder should be able to offer at least the same operational features as the most advanced present day analogue machines – picture-by-picture editing capability, broadcast quality variable slow-motion and still-frame reproduction, 90 minutes playing time, reliable recovery of time code at all speeds, short lock-up time, confidence replay during recording, recognizable picture and sound at shuttle speed, etc. – all these features are expected from any video tape recorder intended for professional use.

The creation of a digital video tape recorder implies, however, the need to solve some problems specific to this form of recording.

The first is the need to record a much larger bandwidth than is required from its analogue counterpart. It is, however, necessary also to keep the tape consumption to a reasonable level (comparable to that of one-inch analogue format type B and C recorders) of about 250 mm/s.

It has already been mentioned that the gross bit-rate required for recording component digital signals is 216 Mbit/s. It is, however, possible to remove from those signals all highly redundant components which can be easily reconstructed at the output – such as line and field blanking intervals (in practice a single digital word is used to indicate line and field synchronization). Such limitations made to the so-called digital active line offer an additional benefit; the bit-rates for the 625/50 and 525/60 standards are virtually identical which, in turn, leads to recorders for different standards having much in common. In practice the entire field interval is not eliminated because in the digital domain a number of ancillary signals and their protection signals have to be carried.

One way to record digital audio on a video tape recorder is to use the same head for both the video and audio signals and to record the audio signals on the same slant track but as separate bursts. The bit-rate corresponding to the audio signals (four audio channels), and the gap between the audio and the video bursts (expressed in terms of a bit-rate) have to be added to the basic (reduced) video bit-rate. Although the overall bit-rate is increased the digital video tape recorder will provide an extremely high quality output. In order to achieve this level of performance it is necessary to protect the audio and video data. Therefore if by adding the audio bit-rate (for four channels), the video and audio error protection overhead, the ancillary signals with their protection and the gap between the audio and video bursts (expressed in Mbit/s) to the net video bit-rate the total bit-rate to be recorded is in excess of 225 Mbit/s.

It is possible to handle this high bit-rate by using parallel channels, but it must be kept in mind that each recording/reproducing channel requires

its own write/read electronics and it own write/read head. In other words each additional channel leads to a more complex machine. Such a high bit-rate, together with the requirement for reasonable tape consumption, leads to the necessity of implementing very high packing densities.

Packing density can be increased either by the reduction of track-width, or by reduction of the shortest recorded wavelength, i.e. record more bits per unit of tape length. Both the track width and the wavelength have direct repercussions on the signal-to-noise ratio. Halving the track width results in a 3 dB reduction in signal/noise ratio, but halving the wavelength results in a 6 dB reduction. This is because halving the track width reduces by two the total volume of magnetized particles, while halving the wavelength reduces by two not only the dimensions along the tape but also the depth of recording thus reducing the total volume of magnetized particles by four.

Two additional factors must here be taken into account. The reduction of the recorder wavelength requires a reduction of the head gap; narrower gaps (besides being more difficult to manufacture) are more sensitive to dust particles and produce longer drop-outs. On the other hand the trackability, i.e. the ability of the head to follow a previously recorded track, becomes more difficult as the track width is reduced. The necessary packing density has to be obtained by means of a compromise between the track width and the minimum recorded wavelength.

The multi-generation capability of a recorder is one of the most important qualities of a given recording format since all post-production is done by making a number of consecutive dubs of the original recording. The limiting factor in the multi-generation process is the fact that the accumulation of errors being relayed from one generation to the next, eventually degrades the picture and the sound so much that they are no longer usable. In the digital domain errors can be corrected and/or concealed. Error correction makes feasible a perfect replica of the input signal at the output, but, since the recorder cannot be considered as a channel with a highly predictable behaviour, full correction would need a massive additional overhead to the basic signal (which in turn means a larger bandwidth). On the other hand, an excellent concealment scheme may facilitate an excellent picture at the output, but the accumulation of concealed errors in a multi-generation process would seriously impair the quality of higher order generations. It is clear that there is a similar requirement for again defining a suitable compromise between the correction and the concealment capabilities which will allow for a manageable bit-rate and an unimpaired multi-generation capability.

1.4 THE STANDARDIZED DVTR FORMAT

It has been seen that digital recording requires a completely new approach, as well as the solving of certain problems which have not been experienced in the domain of analogue recording. Furthermore, the new digital video

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CHAPTER 5
TRACK PATTERN

5.1 INTRODUCTION

Probably the most important aspect of any VTR specification process is to ensure that tapes recorded on different machines will be compatible, since without this a recorded tape may be useless to another organization wishing to use or transmit the material recorded on it. Compatibility can be ensured by either of two methods: one, is to design and specify a standardized tape transport arrangement (upper and lower drum, head and tape guide positions and so on) and let the track pattern naturally result from that design, while the other method is to specify only the track pattern and leave the equipment manufacturers the freedom to design whichever transport arrangement they need. This latter method is the one chosen for the DVTR; it allows a manufacturer to design a machine which is totally optimised for the required task, i.e the maximum number of features, smallest size lowest weight and so on.

The track pattern is thus of foremost importance and decisions on track pattern parameters were made following those concerning the tape and cassettes. The tracks had to allow for the following signals:

(1) One component digital video signal
(2) Four digital audio signals
(3) One time code signal (which needed to accommodate the option of supplying the data for two time codes)
(4) One edit (cue) audio signal in analogue form
(5) One servo control (CTL) signal

The main considerations were that the pattern decided upon should allow broadcastable pictures to be obtained at non-normal speeds, and to permit the recovery of recognisable pictures in shuttle modes when the tape may be travelling at around 40 times normal speed in either the forward or reverse direction. Finally, and of utmost importance, was commonality between different television standards.

In an ideal world all TV standards would be identical and transfer of recorded material to another area or country would present no problem whatsoever. Unfortunately this situation does not exist so the best achievable compromise is to agree on a design that requires the least number of changes, either mechanical or electrical, when operating over different television standards.

The original DVTR development work of the early 1970s had been based around recording of a sampled composite signal; this was natural since the studios and production facilities existing at that time provided a composite environment, e.g. broadcast VTRs, vision mixers, editors, etc. When, in 1979, official standardization discussions started, the broadcasting world had come to appreciate the advantages of the component approach. Also, the definition of the digital component interface standard, to become known as CCIR Recommendation 601, was by now well under way. It was agreed therefore that DVTR development would proceed by employing component video recording using the luminance signal (Y), and the two colour-difference signals ($R-Y$ and $B-Y$).

The advantages of the component approach are many, for example, elimination of luminance/chrominance crosstalk (cross colour/cross luminance), as well as the elimination of PAL, NTSC and SECAM field sequencing due to the subcarrier structure. Most important however is that component signal recording allows a high degree of commonality between TV standards, i.e there is no on-tape difference between recordings made using signals from any of the various colour TV standards that operate at a line frequency of 15625 Hz and a field frequency of 50 Hz (this includes PAL variants *B*, *G*, *H*, *I*, and *N*; and all variants of SECAM).

Having achieved this high degree of commonality, the next step to be taken in the standard definition process was to develop a track pattern that would be as similar as possible for the TV standards operating at field frequencies of 50 Hz (all the PAL variants – with the exception of PAL M – and all of the SECAM variants) and 60 Hz (NTSC and PAL M). The DVTR standard achieves this aim by the use of a segmented format which takes advantage of the 5 to 6 relationship between the two different field frequencies. Segmentation, and other aspects of the format, are described under the headings which follow.

5.2 TRACK PATTERN

Figure 5.1 shows the track pattern; this is identical for both 625/50 and 525/60 systems. The dimensions are the same for both systems except for Program Area Width (F), and Programme Track Total Length (N) as indicated. The reason for these slight dimensional variations is discussed later in this chapter.

Figure 5.1 shows that the track pattern comprises three longitudinal tracks (one each for cue audio, CTL, and time code) and the slant tracks, called Programme Tracks, which contain the digital video and audio data. Each programme track comprises two video sectors and four audio sectors—these are described separately in Section 5.3. The following information details the decisions relating to some of the key parameters of the track pattern.

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5.2.1 Programme track pitch, width and recorded wavelength

All three of these parameters affect the recording density. Ideally the dimensions for all three should be as small as possible so that a high bit-packing density can be realised which, in turn, would ensure low tape consumption. As with most aspects of engineering, however, the ideal is never really achieved because of the need to balance the achievable with the practicable.

It would be possible to use a track width of around 25 μm with a track pitch of the same figure thus giving zero guard band. A narrow track such as this would, however, produce a higher noise figure during playback than that of a wider track and would be more difficult to trace accurately, especially when the requirement for tape interchange between machines is taken into account. A zero guard band would result in tracing an adjacent track while mistracking, thus causing possible data corruption. Similarly a very short wavelength, though ideal, increases the sensitivity to tape dropouts and spacing loss. So, although such format parameters may be viable in a well-controlled laboratory environment, the compromises dictated by practical usage in a wide range of real operating environments, together with the requirement for tape interchanges between machines from different manufacturers, led to the choices of a minimum recorded wavelength of 0.9 μm (more precisely, 0.90549 μm), a track width of 40 μm and a track pitch of 45 μm.

5.2.2 Total data rate and derivation of track length

Track length is derived from the segmented DVTR format design in conjunction with the total data rate to be recorded and the chosen recorded wavelength. The tape format and recorded wavelength are as discussed above and later in this chapter; the total data rate, and the way this is derived, is detailed next. It should be noted that in deriving the total data rate, block and track formatting, audio control and interface data and the amount of data redundancy for error protection need to be taken into account. These aspects will become clear when Chapters 5 to 9 are read, the terms used in the following paragraphs, therefore, are not expanded upon here.

The total data rate is derived from a combination of the following:

(1) The rate of the video and audio information data.
(2) The amount of data added for error protection (redundancy).
(3) The data added to identify the formatted on-tape data blocks.
(4) Audio processor control and interface data.

Video data rate including redundancy and block formatting

The video interface passes data at the rate of 27 Mword/s, which can be serialized to 216 Mbit/s. Because active video data are not present during line and field blanking, and since only 300 lines per field are recorded in 625/50 systems (250 in 525/60 systems), the video data rate is reduced from

216 Mbit/s to 172.8 Mbit/s. This is derived, for the 625/50 system, as follows:

$$1440 \times 8 \times 300 \times 50$$

where 1440 is the number of words in each digital active line; 8 is the number of bits in each word; 300 is the number of lines recorded in each field; and 50 is the number of fields per second.

The equation for the 525/60 system produces an identical result:

$$1440 \times 8 \times 250 \times 60$$

The total recorded video data rate is a combination of the video data rate of 172.8 Mbit/s, given above, and the extra data added for redundancy and formatting. The result is that the video data rate is increased by the equation:

$$172.8 \times (134/120) \times (32/30) = 205.824 \text{ Mbit/s}$$

where 134 is the total number of bytes in a sync block and 120 is the number of infomation (video) data bytes in a sync block; 32 is the total number of bytes in a video outer code block and 30 is the number of video data bytes in a video outer code block.

Audio data rate including redundancy and block formatting

Each audio interface passes data at the rate of 3.072 Mbit/s. This is then doubled to indicate the data rate of all four channels, resulting in a data rate of 6.144 Mbit/s. The source data rate of the bare audio data is derived from:

$$48000 \times 20 \times 4 = 3.84 \text{ Mbit/s}$$

where 48000 is the audio sampling rate; 20 is the maximum number of bits to be recorded; and 4 is the number of audio channels.

The rate, to tape, of this source audio data, is further increased by block formatting; duplication; and the addition of status bits and redundancy. The result is given by:

$$3.84 \times (10/7) \times (168/160) \times 2 \times (134/120) = 12.864 \text{ Mbit/s}$$

where 10 is the total number of bytes in an audio outer code block and 7 is the number of audio data bytes in an audio outer code block; 168 is the total number of data words in an audio product block and 160 is the number of audio data words in an audio product block; 2 accounts for the duplication of data; 134 and 120 relate to sync blocks as discussed for the video data rate

Data rate resulting from edit gaps, preambles and postambles

Each programme track consists not only of the video, audio and other data (discussed above), but also of gaps in which audio edits are made and of pre- and postambles. Together, these add over 6.6 Mbit/s to the data rate which is derived as follows:

$$(10 \times 134) + 30 + 6 \times 600 \times 8 = 6.6048 \text{ Mbit/s}$$

where 10×134 gives the number of bytes equivalent to the five audio edit

gaps and the associated sector pre- and postambles; 30 is the track start premable; 6 is the track end postamble; 600 is the number of tracks per second (24 × 25 for 625/50 systems, or 20 × 30 for 525/60 systems); and 8 is the number of bits per byte.

The *sum* of the three data rates derived in this and the foregoing two paragraphs produces the *total data rate* of:

$$205.824 + 12.864 + 6.6048 = 225.2928 \text{ Mbit/s}$$

Track length

The derivation of track length for 626/50 systems is as follows: For each recorded field there are 4.505 Mbit (225.2928 Mbit/50). For each track (1/12 of one field – see under Section 5.3) there are 375.488Kbit (4.505 Mbit/12). The Synchronized, Scrambled, Non-Return-to-Zero (SSNRZ) channel coding (Channel refers to the path to and from the tape which needs special treatment to ensure a low off-tape error rate – see Chapters 6 and 9) allows two data bits to be recorded in each 0.90549 μm wavelength, thus 2.208 bits are recorded for each μm, the track length therefore is 170 mm (375488/2.208).

The derivation for 525/60 systems produces an identical track length figure. This is because the data rate is the same but only 10 tracks per field are used (see under Section 5.3).

The track length for 525/60 is therefore:

$$225.2928 \text{ Mbit/60 fields/10 tracks/2.208 bits} = 170 \text{ mm}$$

Because the actual field frequency for NTSC is 59.94 Hz and not 60 Hz (carried out to shift the subcarrier harmonics away from the existing sound carrier) the track length becomes 0.1% different from 170 mm. This is shown by dimension N of Fig. 5.1 which gives the 525/60 conversion as 170/1.001. For practical purposes the tracks for both standards can be considered identical. It follows that if the track length dimension is slightly different between 625/50 and 525/60 systems the overall width for the programme area must also be different, this is shown by dimension F in Fig. 5.1.

5.2.3 Data sector dimensions

The dimensions of the video and audio data sectors are derived from the amount of data to be recorded in those sectors. To make sure that small amounts of data can be identified and used when operating at a non-normal playback speed, the data of each sector are arranged in a block form. This block form, which includes the error protection data (redundancy) as well as the video or audio information and block identification data etc., is detailed in Chapters 6, 7 and 8. The parameters and terms given in these Chapters are introduced below to show how the video and audio sector lengths are derived.

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Derivation of video sector length

Each video sector contains 160 sync blocks, one preamble and one postamble. Each sync block comprises 134 bytes of data. The preamble comprises 30 bytes and the postamble, six bytes.

The number of data bytes in each video sector is therefore:

$$160 \times 134 + 30 + 6 = 21476$$

The number of data bits in each video sector is:

$$21476 \times 8 = 171808$$

Since, as previously discussed, there are 2.208 data bits recorded in each micrometre, the calculated length of each video sector is 77.78 mm (171808/2.208).

Derivation of audio sector length

Each audio sector contains five sync blocks, one preamble and one postamble. Since the number of data bytes comprised in each of these sync blocks is the same as for video, the derivation of audio sector length is:

$$(5 \times 134 + 30 + 6) \times 8/2.208 = 2.56 \text{ mm}$$

5.2.4 CTL, time code and cue audio tracks

The signals and arrangement of information on the longitudinal tracks are very similar to those on conventional analogue VTRs, i.e. the cue audio is recorded as a normal analogue signal, time code is recorded using bi-phase-mark coding, and the CTL signal, although having a form which suits the DVTR track structure, is a series of double pulses (pulse doublets). Time code and the cue audio signals are recorded using a.c. bias, whereas the CTL signal is recorded directly without a.c. bias. (This method of recording the CTL signal is similar to that used by SMPTE C-format machines.)

Since the vast majority of processing in the DVTR is digital, it may be considered that the cue audio channel could also have been in digital form. This would, of course, have been perfectly feasible, however the following points should be noted:

(1) Quality of reproduction from the cue audio track is not intended, and indeed does not need to be very high.
(2) The signal from the cue audio track needs to be recovered over a very wide range of playback speeds (on the first machines to be introduced, the digital audio signal will most likely be muted at playback speeds other than normal).
(3) Adopting the digital approach would mean added complexity (and cost).

It was logical therefore that the conventional analogue method was adopted.

Details concerning the information recorded on the longitudinal tracks are provided in Chapter 10.

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5.3 PROGRAMME TRACKS

The programme tracks (Fig. 5.1) contain the digital video and audio data. The total data rate is 225.2928 Mbit/s. Since two data bits are recorded within the 0.905 μm wavelength a bandwidth of around 113 MHz is needed. If the data were to be recorded on a track-per-field basis (as with C-format machines) the track length for 625/50 systems would have to be 2040 mm (113 MHz/50 Hz) $\times$ (0.9×10^{-6}). This would require a drum diameter of 640 mm, clearly too large for practical use. A segmented format, in which each field is recorded on several parallel and adjacent tracks (in a manner similar to the Quadruplex and B-format recorders), was therefore selected for the DVTR. Unlike other recordings from analogue machines using the segmented approach however, digital recordings are virtually free from undesirable head-banding effects.

With 625/50 systems, 12 tracks each of 170 mm in length (see Fig. 5.1 and Section 5.2.2) are required for each field. The number of tracks is derived simply by dividing the total track length of 2040 mm by the required track length of 170 mm. With 525/60 systems the data rate is identical but the field period is less (16.6 ms compared with 20 ms for 50 Hz systems). The total track length is shorter and thus the number of 170 mm tracks is fewer. Total track length is 1700 mm,

$$(113 \text{ MHz}/60 \text{ Hz}) \times (0.9 \times 10^{-6})$$

therefore the number of tracks for 525/60 working is 10, (1700 mm/170 mm).

Figure 5.2 shows a simplified diagram illustrating the segmented field format for 625/50 and 525/60 systems.

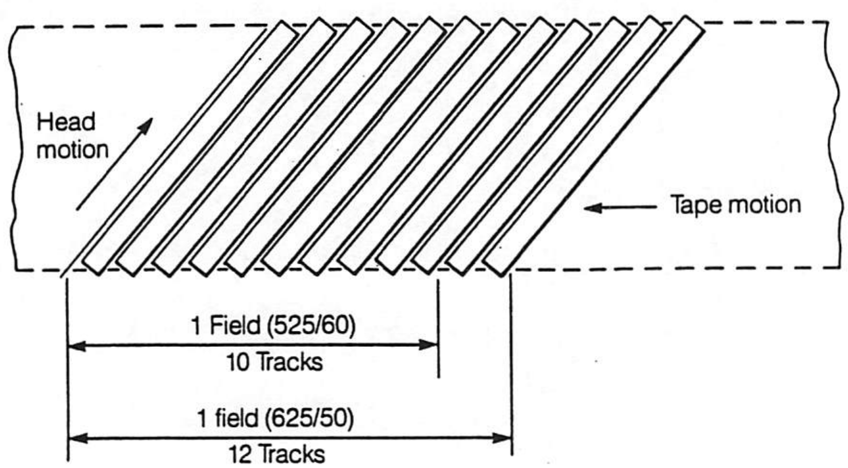

Fig. 5.2 Basic programme track configuration

Within the field segmentation arrangement described, the programme tracks are themselves separated into segments and sectors; this is to facilitate channel processing and the control of off-tape data errors during playback. Details of these aspects are provided in Chapters 6, 7 and 8 but are introduced here, separately for video and audio, to complete the tape pattern information.

5.3.1 Video programme tracks

In order to reduce both the rate and quantity of data to be handled by each processing function, as well as to provide a powerful error concealment strategy, a four channel processing system has been specified for the DVTR (this is discussed further in Chapter 6). The number of tracks in each field has therefore to be divisible by four (without a remainder) to allow field by field recording and editing. With 625/50 systems, having 12 tracks per field, this presents no problem. With only 10 tracks per field in the 525/60 systems, however, division by four is not possible. To overcome this situation each programme track is split into two, thereby creating an arrangement of 24 half tracks in 625/50 and 20 half tracks in 525/60 systems – numbers which, while retaining format compatibility, are divisible by four.

Each half track is called a sector. Four sectors, each of which is recorded using a separate head and associated processing channel, comprise a segment. A segment contains all of the active video information over a period of 50 television lines. The arrangement of video sectors and segments is shown in Fig. 5.3 (see also Fig. 5.1).

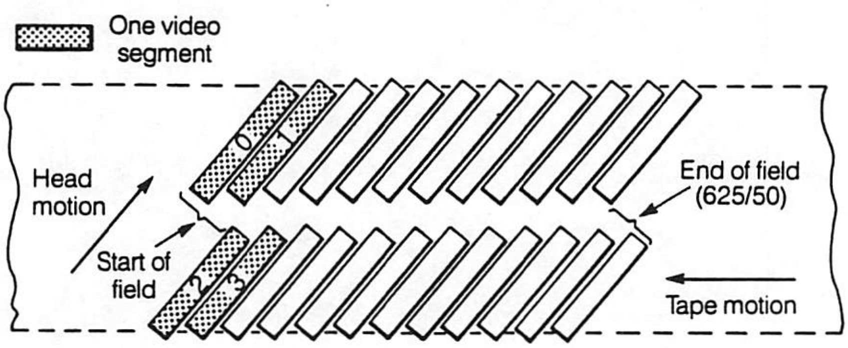

Fig. 5.3 Programme track video sectors and segments

The sequence in which digitized video samples are written to tape together with the reasons that the sectors and segments shown in Fig. 5.3 are so arranged, are described in Chapters 6 to 9.

5.3.2 Audio programme tracks

In order to enable common data channels and heads to be used for both audio and video data it was always intended by those involved with DVTR standardization that the audio tracks would be multiplexed with the video tracks. Before the video tracks were split in half, the intention had been to place the audio data at the track ends, either at one end of the track or, to improve data error control, at both ends. It was considered, however, that tape edge damage and tape skew errors could introduce problems with these methods so it was fortunate that the video tracks were split since this allowed audio data to be located in the centre of the tracks between each video sector.

Figure 5.4 is a simplified diagram which shows the result of locating the audio data in the centre of the track. From this diagram it can be seen that

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the central location not only offers better protection from tape edge damage but also suffers much less from mis-tracking, as a result of skew error, than the alternative locations previously described. This location is advantageous also because the audio data occupy a far smaller portion of the track than the video and, correspondingly, are more vulnerable.

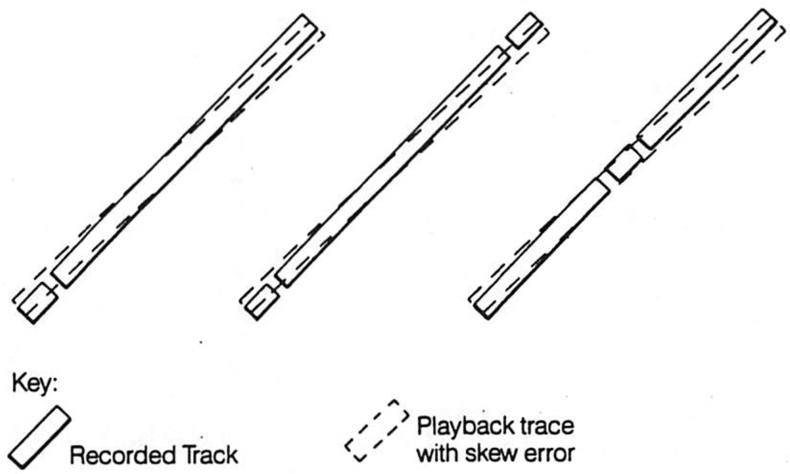

Fig. 5.4 Audio position options

Because the ear, unlike the eye, is very critical to errors and disturbances, the control of errors in the audio signal is of paramount importance. Audio error control is dealt with in Chapters 6 and 8 but the on-tape result is that sectors and segments are introduced in a manner similar to those of the video programme tracks previously described.

One audio segment contains digital audio data from one audio channel over a 6.66 ms time period, this is equivalent to one sixth of a frame for 625/50 and one fifth of a frame for 525/60 systems. Each segment comprises four audio sectors distributed among four adjacent programme tracks. Each of the four sectors contains either odd or even audio samples (together with processor control and redundancy words for error correction). Two of the four sectors, therefore, provide all the necessary information for the 6.66 ms time period; the two additional sectors contain duplicate data to enhance the power of the error correction system.

The method adopted for sector distribution within a segment ensures that each sector is:

(1) Recorded at a different distance from any one edge of the tape.
(2) Recorded by each of the four separate processing channels and heads.

This method ensures that in the event of a data error caused either by a tape dropout burst, longitudinal scratch or head clog, recovery or generation of the correct data can reliably take place at normal playback speed.

Figure 5.5 illustrates the arrangement of audio sectors and segments in an expanded form to aid clarity (see also Fig. 5.1). The diagram highlights the segment of audio channel one. Note that the spaces left between each

2<s>8</s>6

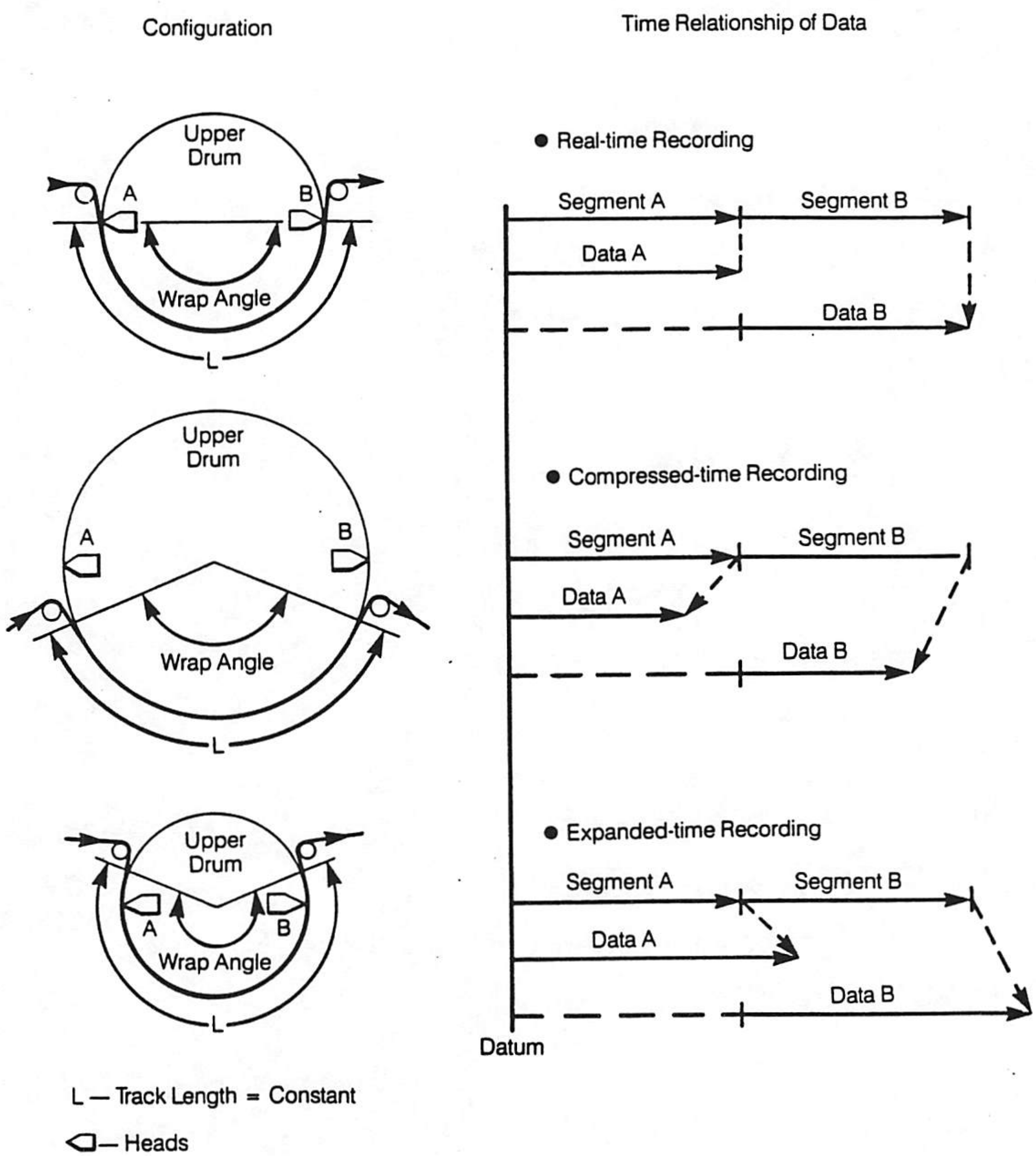

Fig. 5.7 Examples of compatible drum configurations

5.4.2 Drum (head) rotation rate

The rotation rate of a drum is determined by the field rate, the number of tracks per field, and the number of heads used to record those tracks. This is true for all VTR formats. Taking the non-segmented C-format as an example, where just one head is used to record one track for each field, drum rotation rate is equal to the field rate. For the 625/50 system this results in a rotation rate of 50 rps, or 3000 rpm; while for the 525/60 system the rotation rate is 60 rps, or 3600 rpm. Taking the non-segmented U-format as another example, the drum rotation rate for the 625/50 system works out at just

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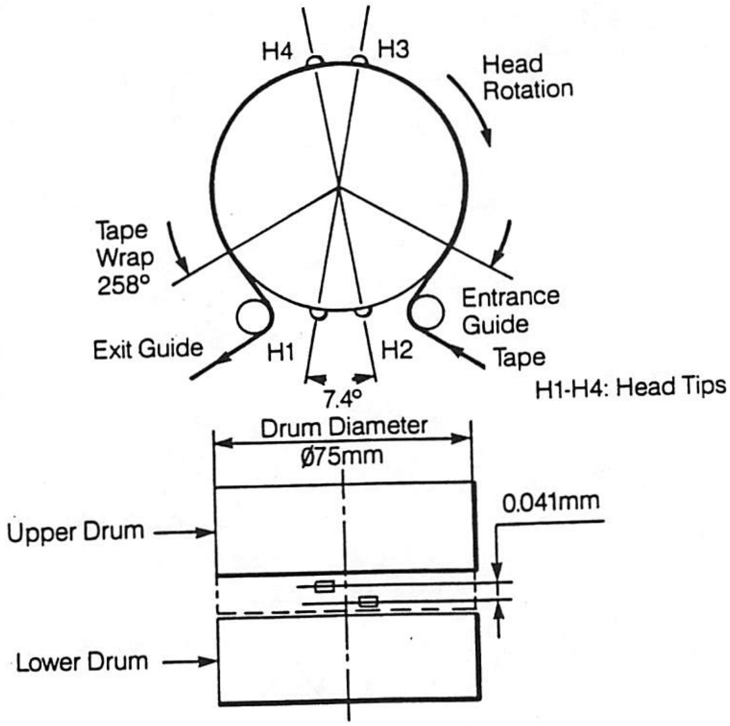

Fig. 5.8 Drum configuration I

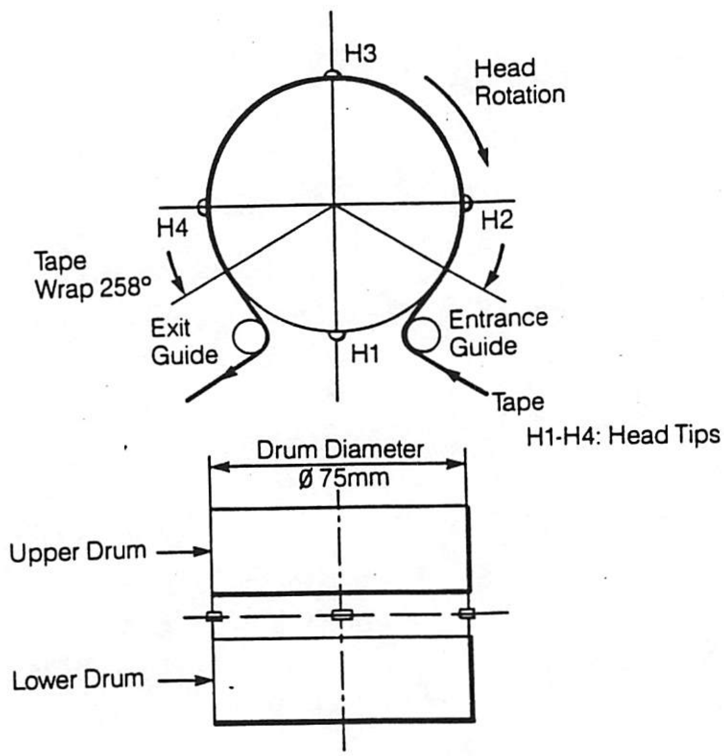

Fig. 5.9 Drum configuration II

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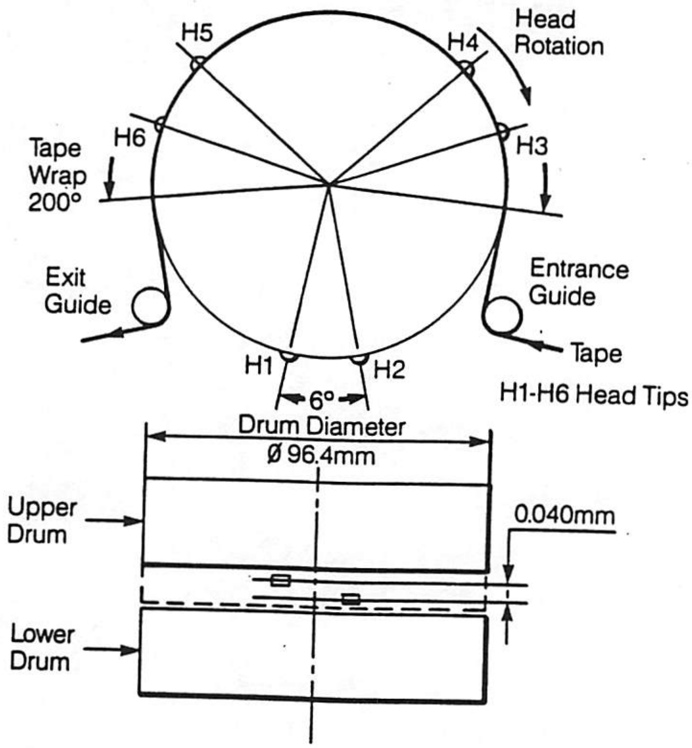

Fig. 5.10 Drum configuration III

Table 5.1 HEAD DRUM PARAMETERS

Parameter	*Model*		
	I	*II*	*III*
Head rotation rate	150 rps (150/1.001)*	150 rps (150/1.001)	100 rps (100/1.001)
Drum diameter	75 mm	75 mm	96.4 mm
Head to tape speed	35.34 m/s (35.34/1.001)	35.34 m/s (35.34/1.001)	30.15 m/s (30.15/1.001)
Helix angle (degrees)	5.4444	5.4441	5.4517
Minimum number of heads	4	4	6

* The 1.001 figure shown in the table results from the 0.1% difference of field frequency in the 525/60 system (59.94, not 60 Hz).

25 rps, or 1500 rpm. This is half that of the previous example because two heads are used alternately to record each field. The formula for calculating drum rotation rate is:

$$\text{Field rate} \times \frac{\text{Number of tracks per field}}{\text{Number of heads}}$$

which, for U-format operating in 625-line mode is 50 × (1/2).

In the 4:2:2 DVTR application, there are 12 tracks per field when used in the 625/50 mode of operation. If, as an example, only one head were used (remember that in practice at least four heads must be used), the rotation rate of the drum would be 600 rps, or 36000 rpm (50 Hz × 12 tracks).

Using a drum containing the minimum of four heads to record the same number of tracks produces a drum rotation rate of 150 rps, or 9000 rpm (50 Hz × (12 tracks/4 heads)). This is the rotation rate of drum types I and II shown in Table 5.1.

It follows, therefore, that using the Type III drum which has six heads the rotation rate works out to the lower figure of 100 rps, or 6000 rpm.

Because of the 5-to-6 ratio of both the field rates and number of tracks, the drum rotation rate to make 4:2:2 recordings using the 525/60 system is the same as that used for 625/50 recordings, i.e.

$$60\text{ Hz} \times (10\text{ tracks per field}/4\text{ heads})$$

Of course the small 0.1% difference due to the 59.94 Hz field rate instead of 60 Hz remains.

5.4.3 Data rate to rotary heads

The data rate to each rotary head is determined by the tape wrap angle, the number of heads and the original source data rate to be recorded. The value for the source data rate is fixed at 225.2928 Mbit/s, this is detailed in Section 5.2.2. The wrap angle and number of heads can, of course, be changed, and so become the variable aspects of the complete data rate formula which is:

$$\text{Source data rate} \times \frac{\text{Maximum wrap angle}}{\text{Actual wrap angle} \times \text{Number of heads}}$$

where the maximum wrap angle is 360°.

From this it may be appreciated that the actual data rate to the heads is lower if either a larger wrap angle is employed or the number of heads is increased or, of course, both.

If, as an example, only one head were to be employed with a 360° tape wrap (in practice such a scheme cannot be used), the data rate fed to that head would equal the source data rate of 225.2928 Mbit/s.

Realistically, however, using the above expression the data rate to each head of a type I or II drum (see Table 5.1) works out to 78.59 Mbit/s; while for the type III drum, the rate to each head is 67.58 Mbit/s.

5.4.4 Rotary data and erase head arrangements

Data (video/audio) record/playback heads

As previously mentioned, all DVTRs must have at least four rotary heads to record or playback the digital information. The heads may be made of ferrite or other suitable material which is able to satisfy the necessary parameters.

Zero azimuth recording is used by the DVTR for the programme tracks; the azimuth of all the head gaps used to produce the recorded tracks is, therefore, perpendicular to the tracks themselves. This enables pictures to be recovered in variable and shuttle modes by allowing the heads to pick up individual bursts of video data from adjacent tracks. Second generation machines may access audio from the digital programme tracks when in these variable modes. First generation machines, however, due to the complexity and cost of the extra circuits needed, will use only the longitudinal cue audio track when operating at non-normal speed.

The mechanical arrangement of the heads and drum will be decided by individual manufacturers; the heads may form part of the upper (rotating) drum or, be arranged on a separated head disc located between the upper and lower parts of the drum. Signal flow between the heads and the record/playback amplifiers of the four processing channels may be achieved in the familiar way using rotary transformers, but other methods may be used to fulfil design requirements.

Although the minimum number of heads needed on a DVTR drum is four (the same four being used for both record and playback), many more may be used. For instance, to provide off-tape playback during record (confidence playback) an additional four heads are necessary. Another possible facility is to allow audio data on one track to be read, then output in either digital or analogue form to an external unit for processing (e.g. for equalization or mixing with other material) and then returned to the same track (or another track) in-phase with the original material. This process is called Read-Modify-Write and to do this, another four heads are needed to read the on-tape data in advance of that read by the normal heads.

It is most likely that all heads on the first generation of machines will be fixed (rigid); that is, there will not be any dynamic or automatic track following heads for stunt playback modes. The reason will simply be lack of development time and the additional complexity of the head drum and electronics. The problems to be overcome are:

(1) Space – the amount of space available in the smaller drum sizes (75 mm) when all the other heads are installed is not very great. Also, because of the bimorph assembly (the arrangement of the piezo-electric element which is used to move the head), dynamic track following heads are much larger than normal heads so compounding the problem.
(2) Drum rotation rate – the drum rotation rate is typically 150 rps (9000 rpm) which is three times faster than the equivalent C-format drum; clearly, the forces to be overcome by the head servo system are that much greater in the DVTR. Another aspect of high drum speed is, of course, that the time available in which to overcome these forces is very short.

These problems will take some time to solve but will, no doubt, be overcome in the future. In the meantime the result is that broadcast-quality stunt modes from studio machines will be limited to a range of slow speeds, perhaps in the region of still plus or minus 0.25 times normal speed. Outside

of this speed range the heads cannot read all the information on the tape and so a broadcastable picture is not possible; the resulting picture will, however, be clearly visible (and in colour) right up to about 40 times normal speed.

Data (video/audio) erase heads

To ensure that all data bits on the programme tracks are recorded on-tape without corruption by previously-recorded material, the tape needs to be erased. A full-width erase head may be used for normal (crash) record, but to allow insert and assemble editing to be carried out the DVTR must use some form of rotary (or flying) erase heads in a similar way to that presently found on analogue VTRs. When it is considered, however, that a studio-type machine may need 12 or more head assemblies, the overhead of an additional four separate heads for erasing the programme tracks needs to be avoided wherever possible. This can be accomplished by the use of heads with double gaps, i.e. an erase gap and a record gap. In this way the record heads provide the erase as well as the record function.

The Evolution of Video: New Standards and Image Quality

YVES FAROUDJA

When I was in high school (more precisely *Lyceé*) in Paris many, many years ago, I heard that CBS was demonstrating its color television system (which was the U.S. standard at the time) in a hotel somewhere on the Right Bank.

So I went to see it, and it was beautiful. A large disc was rotating in front of the CRT and the colors were absolutely perfect, precisely matching the original subject (a young lady with colorful balloons), which was in the next room.

And then it was abandoned and NTSC was selected. Why? Because a standard is not an arbitrary edict imposed by a governmental or industrial organization. A good standard is a codification, after the fact, of what Nature imposes and what people are beginning to do of their own free will. The fact is that in 1953 mechanical displays of television signals were gone, with Mr. Baird and Mr. Nipkow.

Compatibility with black-and-white broadcasting was paramount. Nature did not want a TV set with moving parts, and RCA won with a compatible system. The glorious, fully electronic NTSC system was selected, followed a few years later by its pale imitators (PAL and SECAM, which were presented as improvements). NTSC is still alive and well 43 years later.

Now we are on the verge of another abrupt change. Little by little, digital technology has invaded different areas of the video engineering field, and it's probably time for a change.

It's time for a change anyway, because people apparently don't like to do the same thing for too long a time, even if it works. So, if we do have to change, what does the digitization of television bring to the party?

To understand the answer it's interesting to follow the workings, goals, and decisions of two different committees and several organizations' efforts. The first one is the HDTV effort, the second is the MPEG/JPEG committee work.

The HDTV work, which started in Japan in the early Seventies, cannot be called a success. Its original tenet is that NTSC and PAL are incapable of delivering an image of high quality to the home viewer, not having enough lines (525 or 625), and being heavily loaded with ugly artifacts. HDTV (or ATV—advanced television—as it's now called in the U.S.) turned digital circa 1990, following General Instrument's lead, but is still based on the belief that the key to image quality is to broadcast more lines.

On the other hand, MPEG guidelines are vague and indefinite and apply to whatever number of lines you want to use. These guidelines appear to be more like a set of grammatical rules than a set of precise numerical values for the system parameters. For example, MPEG does not say: Your subcarrier frequency shall be 3.579545 MHz (as does NTSC). Instead MPEG says: You shall use certain techniques (DCT, motion vectors, etc.) to compress video. MPEG does not impose very specific numerical goals. Its main objective is to save bandwidth. This is a noble cause, as bandwidth is a sparse, God-given, very precious natural resource that's not to be wasted.

Now, why is MPEG right? Why is it to be adopted by DSS, DVD, (Digital Satellite Systems, Digital Video Disc), etc? It's simply because it's in tune with today's technology. The standard can be vague if the receiver can be programmed to follow the different embodiments of the signal sent by the source.

Advances in software engineering allow us to consider a video source as a "magma" of bits: exploitable, manipulatable, and displayable at mercy. It would not have been possible 20 years ago. It will be even easier 20 years from now.

We are witnessing a slow disappearance of the old analog video processes. This is not imposed from the top (such as HDTV) but from a natural, bottom-up, seemingly effortless evolution, which seems to go, on its own, in the right direction without an ukase from the Czar, or a Revolution.

First there's the studio. The analog camera or film chain signal is quickly digitized because it is easier, because it is better. It's often, unfortunately, manipulated in NTSC (because it is there), and then redigitized. The signal is then compressed, broadcast via satellite, (or soon to be recorded on tape or disc), then decompressed. Today, however, the component signal is still remodulated in NTSC and RF in order to be displayed on existing sets.

The future evolution of signal processing is evidently not too difficult to imagine. Clearly, the last bastions of analog processing should fall. First, there is no reason not to go directly

(continued on page 214)

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Section 3

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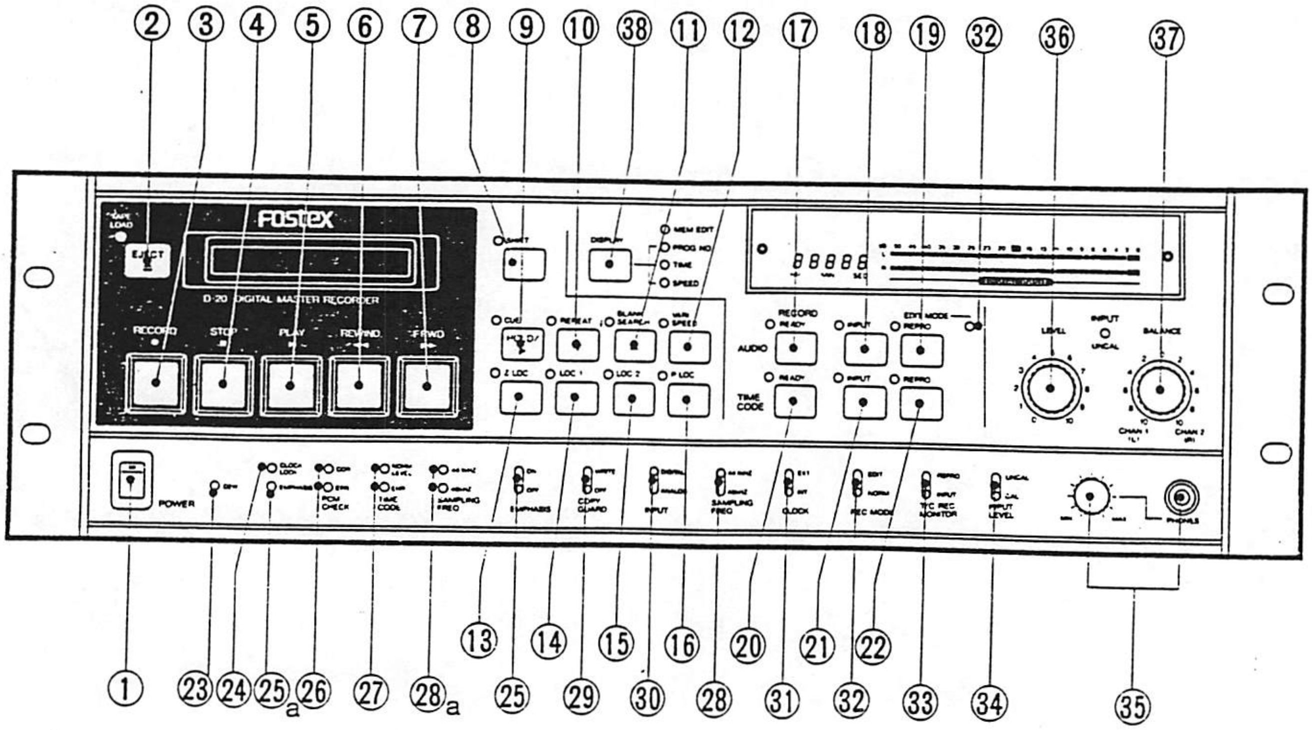
CONTROLS AND FUNCTIONS

FRONT PANEL

Figure 10

Front panel view



The front panel of the D-20 can be thought of as four separate areas; the darkened area where the main tape handling controls are located, the middle area where the controls are mainly for locating and editing operations, the area on the right where most of the recording and edit controls are located, and the panel of toggle switches on the bottom where tape information and formatting is done. The D-20 uses an ingenious shift key enabling eight keys to serve fifteen functions. Please study the following material and we think you will soon see how easy the D-20 is to operate.

1. POWER switch

2. EJECT button

Press to eject the DAT cassette. The **EJECT** button only works in the stop and pause mode. When the tape is being loaded or unloaded by the cassette transport, the **TAPE LOAD** LED will blink. When there is a tape in the

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machine, this LED will remain lit.

3. RECORD button

When the **AUDIO READY** key or the **TIME CODE READY** key are ON, pressing this button and the **PLAY** button puts the unit in the record mode. The **RECORD** button will light red.

4. STOP button

The **STOP** button has two steps in its function. Pressing this button once puts the transport into the pause mode. The **STOP** button will light while the **PLAY** button will blink. Pressing this button twice will put the transport into full stop mode. Only the **STOP** button will be lit.

< Note >

The pause mode is incorporated to reduce start up time. However, the transport will enter full stop mode if the pause mode is engaged longer than three minutes. The reason for this is to remove the tape surface from contact with the rotary head drum and protect the tape from premature tape wear. Each push of the **STOP** button will alternate the transport from stop to pause mode.

5. PLAY button

Pressing this button puts the unit in playback mode; the **PLAY** button will light. When pressed with the **RECORD** button it also serves to activate the recording mode. The **PLAY** button also will engage other functions such as repeat and locate play; these will be explained latter.

6. REWIND button

When this button is pressed once, the tape is rewound at five times normal speed and the **PLAY** and **REWIND** buttons are lit. Pressing the button once more will activate high speed winding (100 times normal speed). Only the **REWIND** button will be lit. Each press of the **REWIND** button will cause the machine to alternate between high and low speed rewinding.

7. F FWD button

The **FFWD** button operates in the same manner as the **REWIND** button.

8. SHIFT key

Keys 9 through 16 have dual functions. Pressing the **SHIFT** key engages these separate functions as will be explained below. The **SHIFT** LED will

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light when the **SHIFT** key is engaged. Each push of the **SHIFT** key alternates between the shift and non-shift mode.

9. HOLD/ ➤ /CUE key

Non-shift mode: Depending on the current operation, the key serves to hold the display value or to move the edit point. Pressing the **HOLD** key and then **LOC 1** or **LOC 2** will enter these time points into the D-20 memory.

Shift mode: Pressing this key activates the CUE mode.

10. ▼ /REPEAT key

Non-shift mode: The ▼ key reduces the values in the MEM EDIT and VARI SPEED modes. Each push of the key decreases the value of the two functions. This decrease will be confirmed on the display.

Shift mode: Pressing this key activates the REPEAT mode for automatic playback between LOC 1 and LOC 2. The **REPEAT** LED will light up.

11. ▲ /BLANK SEARCH key

Non-shift mode: The ▲ key increases the values in the MEM EDIT and VARI SPEED modes. Each push of the key increases the value of these two functions. This increase will be confirmed on the display.

Shift mode: Pressing this key activates the BLANK SEARCH mode. The **BLANK SEARCH** LED will light.

12. VARI SPEED key

Non-shiftmode: Reserved.

Shift mode: Pressing the key activates the VARI SPEED mode. The **VARI SPEED** LED will light up and the display will show the current value in percent with normal speed represented as 0 0. Tape speed can be varied plus or minus ten percent with the ▲ and ▼ keys.

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13. Z.LOC (ZERO LOCATE) key

Non-shift mode: Pressing this key locates the OO HR OO MIN OO SEC point on the tape. If pressed when the unit is in MEM EDIT mode and the display is switched to TIME, the current counter point is memorized as 0:00:00. To escape from Z.LOC press the **STOP** button.

Shift mode: Pressing this key will cause the display to show A-time (absolute time) with the Z.LOC point as reference.

14. LOC 1 key

Non-shift mode: Pressing this key locates the point on the tape that has been memorized as LOC 1. If the unit is in MEM EDIT mode and the display is switched to TIME, the current display is memorized as LOC 1. If the display is in PROG. NO. mode, the displayed program number is located.

Shift mode: Pressing the key causes the time memorized as LOC 1 to be displayed.

15. LOC 2 key

The **LOC 2** key works the same as **LOC 1**.

< Note >

Both of these keys are LED indicated. Also, it should be noted that these keys are not sequential in action. In other words, you may program 1-2 or 2-1 and the tape will locate in the same way. For a better understanding of these keys, please read the item "Repeat Function", and "HOLD key".

16. P.LOC (Play Locate) key

Non-shift mode: Pressing the key locates the time point on the tape that was memorized as P.LOC. This is the point where a PLAY or REC-PLAY command was last input. If the unit is in the MEM EDIT mode and the display is switched to TIME, the current display counter time is memorized as P.LOC.

Shift mode: Pressing the key will cause the display to how the time memorized as P.LOC to be shown.

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< Note >
These locate functions are LED indicated. To escape, press **STOP** button.

17. AUDIO READY key

Pressing this key activates the AUDIO READY mode. The **READY** LED will light. Pressing this key once more activates audio mute (for blank recording). The **READY** LED will flash in this mute mode. Pressing the key once more turns the AUDIO READY mode off.
In the normal mode (**REC MODE** switch set to NORM), the TIME CODE ON/OFF status is also controlled by this key and the **TIME CODE READY** LED will light accordingly.

18. AUDIO INPUT key

When this key is pushed the audio input signal is sent to the AUDIO and the MONITOR output connectors. The **AUDIO INPUT** LED will light.

19. AUDIO REPRO (Audio Reproduction) key

When this key is pressed the tape playback signal is sent to the AUDIO and MONITOR output connectors. The **REPRO** LED lights.
In the edit mode (**REC MODE** switch set to EDIT), the audio input signal is sent to the output connectors as soon as recording is started. In this condition the **AUDIO INPUT** and **REPRO** LEDs are both lit.

20. TIME CODE READY key

Pressing this key activates the TIME CODE READY mode. The **READY** LED lights. Pressing this key once more turns the mode off.
In the normal mode (**REC MODE** switch set to NORM) the AUDIO READY ON/OFF condition is also controlled by this key and the **AUDIO READY** LED lights accordingly.

21. TIME CODE INPUT key

When this key is pressed a TIME CODE INPUT signal is sent to the TIME CODE OUTPUT connector.

22. TIME CODE REPRO (Time Code Reproduction) key

When this key is pressed the time code read from the tape is supplied at the TIME CODE OUTPUT connector. When recording is carried out in the edit mode (**REC MODE** switch set to EDIT) with the **TIME CODE REC**

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MONITOR set to REPRO, the playback time code read from the tape is supplied to the TIME CODE OUTPUT connector (time code refresh).

23. DEW LED

This LED will flash if there is any moisture or condensation on the head drum. If this LED is flashing a tape cannot be inserted into the D-20. If there is a tape in the machine and condensation is detected the LED will flash and all functions will stop until this moisture has evaporated; that is, if the transport is in operation and moisture is detected, the **DEW** LED will flash and the transport will enter the STOP mode. If this condition is encountered, leave the power on as a built in heater is activated when this LED is lit. When the moisture is evaporated, the LED will extinguish. You may then resume operation.

24. CLOCK LOCK LED

When the D-20 is controlled by an external clock (from EXTERNAL SYNCH or DIGITAL INPUT) this LED indicates the signal condition as indicated below:

Input mode \ Clock	Proper	Improper	No input
DIGITAL IN	Light	Blink	Blink
EXT SYNC	Light	Blink	Extinguished

Figure 11
CLOCK LOCK LED indication

When the internal clock is used, the LED lights constantly.

25. EMPHASIS switch
25a. EMPHASIS LED

When recording from the analog input, this switch turns the emphasis circuit on and off. When emphasis is activated, the LED lights .
When recording from the digital input and during playback, the position of the switch has no effect. De-emphasis during playback is chosen automatically. If the emphasis condition of the signal differs from the switch condition, the LED flashes.

26. PCM CHECK LEDs

This pair of LEDs serves to monitor the number of errors in the audio and sub code signals during playback. It also serves to monitor error conditions during recording in the normal mode (**REC MODE** switch set to NORM), when the **AUDIO INPUT** key is set to ON. When a blank tape is played back,

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both LEDs are lit.

Both LEDs off :	Normal playback
COR (green) lit :	Errors occur, but are within the correctable range.
COR (green), **ERR** (red) lit :	Number of errors is too high to allow full correction. Data interpolation and/or muting is applied.

27. TIME CODE LEDs

This pair of LEDs serves to monitor the time code signal in the input or output.

NORM LEVEL (green) lit :	Time code signal level is normal.
ERR (red) lit :	Time code signal format is not normal.

< Note >

An error condition is diagnosed when there is no synch signal.

28. SAMPLING FREQUENCY Switch

28a. SAMPLING FREQUENCY LEDs

The **SAMPLING FREQUENCY** switch selects either44.1 kHz or 48 kHz. The **SAMPLING FREQUENCY** LEDs indicate sampling frequency during playback mode. If the sampling frequency of the signal read from the tape differs from the switch position, the LEDs will flash and no audio signal is output.

Depending on the digital switch settings of the D-20 (for broadcasting, studio or consumer use), the action of the LEDs in the DIGITAL IN mode differs.

< Notes >

In the DIGITAL IN mode (**INPUT** switch set to DIGITAL), when the switch setting and the sampling frequency of the signal being input differ, the signal will not be recieved. To confirm this, please note that the **DIGITAL INPUT** indicator on the meter display does not light. However, when the clock is synchronized, (D-20 will be in vari-speed mode), the input signal will be recieved.

When the DIGITAL INPUT signal is in the AES/EBU format, the LED will blink indicating the D-20 reads the status of the digital signal being input. When the DIGITAL INPUT signal is for consumer use format, the LED will not indicate by blinking.

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29. COPY GUARD switch

Set this switch to WRITE when you wish to record a copy inhibiting code to the tape.

30. INPUT switch

This switch selects either the DIGITAL or ANALOG input. When the digital input is selected and a correctly chosen digital audio signal is supplied at the input, the indication DIGITAL INPUT appears on the meter display. To learn more about the digital audio interface, pease refer to Appendix.

31. CLOCK switch

This switch selects which clock is used by the processing circuit of the D-20.

INT : *Internal clock*
The exception is when the digital input is chosen; the D-20 then automatically switches to external synchronization.

EXT : *External synchronization*
The exception is when no external synchronization signal is present, the D-20 automatically switches to its internal clock. In that case, the **CLOCK** LED goes out.

< Note >
If the input signal is a video signal, lock-up time will be about 90 seconds. In this case, please continue the input of house sync signal.

32. REC MODE switch

This switch selects the the recording mode.

EDIT : This mode serves for editing (punch in/punch out recording). The leading heads are used for playback and the trailing heads are used for recording.

NORM : This is the mode for normal recording while confidential monitoring. The leading heads are used for recording and the trailing heads are used for playback.

33. TC (Time Code) REC MONITOR switch

This switch is only active when EDIT in the recording mode is selected with

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the **REC MODE** switch. This is used for refreshing the time code.

REPRO : The time code signal already recorded on the tape is read and supplied at the TIME CODE OUTPUT connector when the **TIME CODE REPRO** key is pressed. This is used to rerecord the time code in synch with time code information from the tape.

INPUT : When the **TIME CODE INPUT** key is pressed the input signal time code is supplied at the TIME CODE OUTPUT connector. The time code output is the same as that of the audio signal input.

34. INPUT LEVEL switch

This switch selects the level of the analog input signal, calibrated (CAL) or uncalibrated (UNCAL). When the UNCAL position is chosen the **UNCAL** LED will light.In the CAL position an input signal level of +4 dBm corresponds to a level meter indication of -18 dB below the full-scale point.

35. PHONES jack and control

Stereo headphone jack and level control.

36. INPUT LEVEL control

This control adjusts the analog input signal level when the **INPUT LEVEL** switch is set to UNCAL. This control changes channels 1 (L) and 2 (R) simultaneously.

37. INPUT BALANCE control

This control adjusts the balance between the two channels when the **INPUT LEVEL** switch is set to UNCAL.

38. DISPLAY key

Pressing this key cycles the display through three functions. The respective LED will light to indicate the chosen function:

•PROG NO (program number)
•TIME (time information read from the tape)
•SPEED (current tape speed)

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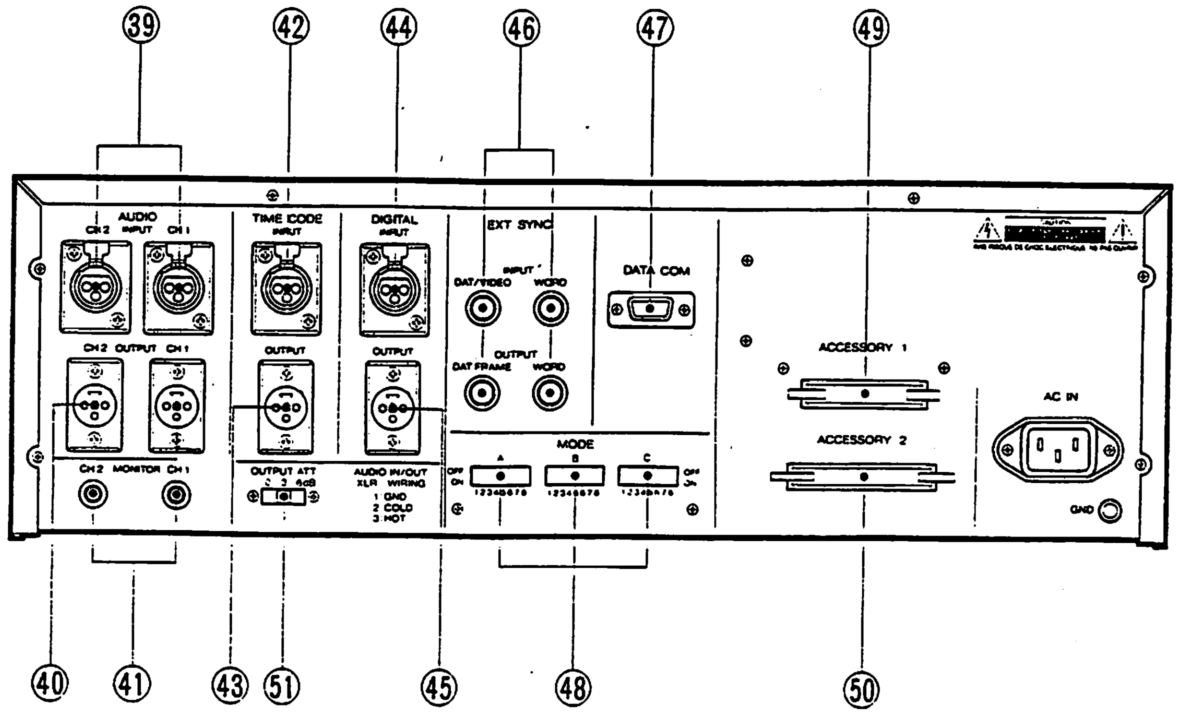
Pressing the **HOLD/▶** key activates the MEM EDIT function. Data being edited are shown on the display counter. The **MEM EDIT** LED will light. Press the **DISPLAY** key to cancel this mode.

** Please refer to section 7, "1. Display Functions".*

REAR PANEL

Figure 12
Rear panel view



39. AUDIO INPUT connectors (CH1,CH2)

Balanced input connectors (XLR 3-31) for analog signal input.

40. AUDIO OUTPUT connectors (CH1,CH2)

Balanced output connectors (XLR 3-32) for analog signal output.

41. MONITOR jacks (CH1,CH2)

Unbalanced output connectors (1/4 inch phone jacks) for analog signal output.

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42. TIME CODE INPUT connector

Balanced input connector (XLR 3-31) for input of SMPTE/EBU time code signals.

43. TIME CODE OUTPUT connector

Balanced output connector (XLR 3-32) for SMPTE/EBU time code signals.

44. DIGITAL INPUT connector

Balanced input connector (XLR 3-31) for digital signal input.

45. DIGITAL OUTPUT connector

Balanced output connector (XLR 3-32) for digital signal output.

46. EXT SYNCH jacks

[INPUT]

DAT/VIDEO : Sync signal (DAT frame, composite video, video frame, video field, etc.) is input.
WORD : WORD SYNC signal is input.

[OUTPUT]

DAT FRAME : DAT frame signal is output.
WORD : WORD SYNC is output.

< Note >
When synchronizing two D-20s, sync operation of word clock accuracy is made possible by parallel use of DAT frame signal and word sync signal.

47. DATA COM connector

9-pin data communication connector (RS-422A)
will be corresponded to SONY 9PIN PROTOCOL.

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48. MODE select digital switches

These switches control settings such as the frame frequency and synchronization mode.

Down position is ON and up position is OFF.

The setting of these switches is read only once when the power is switched on. Change these settings while the unit is turned off, or turn the unit off and on again after changing the position of the switch.

49. ACCESSORY 1 connector

Parallel remote control connector for use with the Fostex 4030 synchronizer.

50. ACCESSORY 2 connector

I/O connector for use with expansion bus.

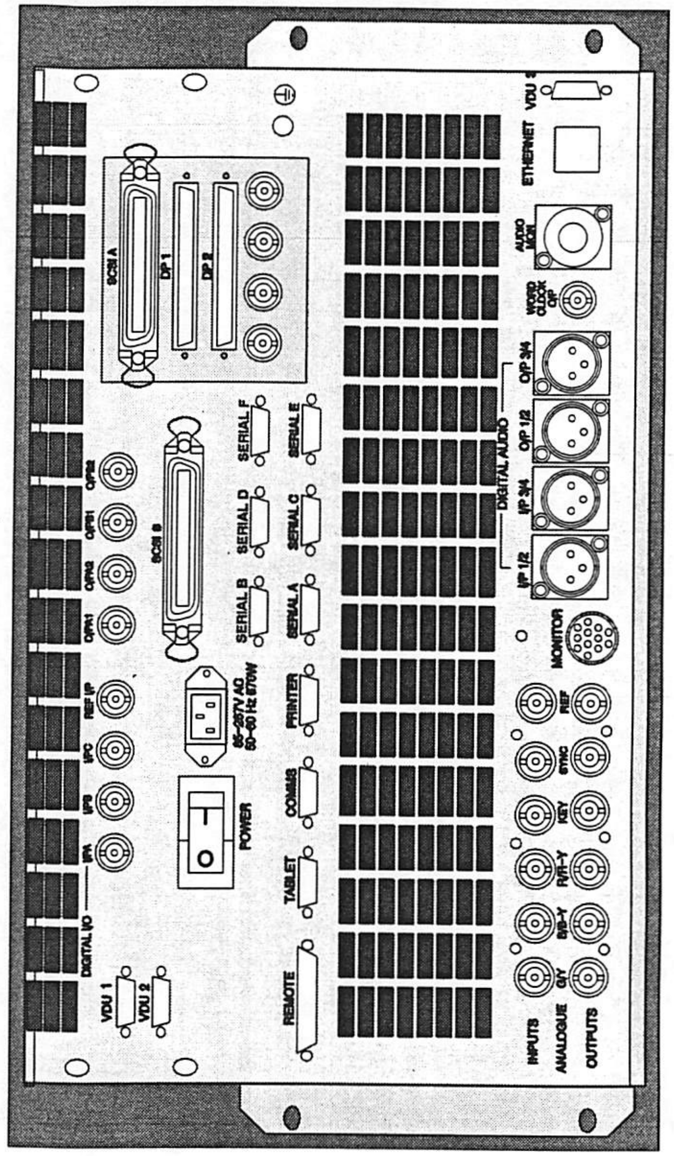
51. OUTPUT ATT (Output attenuator)

Signals at AUDIO OUTPUT (XLR 3-32 OUT) can be attenuated in three steps (0 dB, 3 dB, 6 dB by a switch). Reference output levels are +4 dBm, +1dBm, and -2dBm each.

Summary

We hope the previous section has answered any questions you may have had about the control layout of the D-20. If you haven't already done so, now would be a good time to install your new machine and power up. The following is written on the assumption that you are operating your D-20 in the configuration for which you purchased it.

Rear Panel - Model 6 System



bing, mixer inputs feed tape tracks, and tape tracks feed tape monitor inputs.

Then, as long as you know how to route mixer inputs to tape tracks (via busses or direct output/assigns), and know which tape monitors to listen to and where to find them, you'll be able to record and to hear whichever tape tracks and inputs you want.

As mentioned earlier, many in-line consoles have their group outputs—as well as their tape monitor controls—located within channel inputs. That is to say, group buss 1 is in-line with input channel 1, group buss 2 is in-line with input channel 2, and so forth. Since each input module is both an input and an output, this is often known as an *I/O* design.

Many split consoles have the advantage of being able to use the tape monitor channels as extra line input channels during mixdown (some in-line consoles can be configured to do this, but not as easily). This technique is described on page 99.

While in-line consoles are a bit trickier to understand, they are physically smaller than equivalent split consoles, and are just as easy to use once you get the hang of them.

* * * *

There is a wide variety of mixers and recorder/mixers on the market, and many of us will undoubtedly graduate to bigger and better mixers as we progress. For these reasons, this chapter has emphasized concepts, rather than rote, step-by-step instructions.

We've seen the three main sections of multi-track mixers: the input, output, and monitor sections. We've covered many of the fundamental components you can expect to find in all of those sections—just remember that the three main sections of multi-track mixers can be found in a wide variety of different physical permutations. These variations can apply to any mixer or recorder/mixer of any size and though some may appear very different from one another, most are functionally similar.

We'll see more of mixers in action in the coming pages. When we start to lay down tracks in Chapter Twelve, be sure to refer back to this section anytime you have a question about mixer terminology or concepts. Being able to understand and use your mixer creatively may at times feel like a game—but it needn't remain a puzzle! □

CHAPTER 8: SIGNAL PROCESSING

Signal processing is to the recording engineer as lighting is to the photographer. It allows the subject—music—to be colored and expressed in a variety of ways. Used properly, it can enhance your music and offer more professional-sounding results. But in the same way that the best lighting can be of little help to a photographer with a poor choice of subject, all the latest and greatest signal processing gear is only a tool, and doesn't guarantee good music. In fact, many of the best producers realize that an overkill of effects can ruin a good song.

In any case, the last few years have seen the cost of effects and other signal processing gear drop to new lows, and a lot of this gear can be extremely useful to enhancing your music—just as lighting can enhance a photograph's mood. Effects that until a few years ago were reserved for the elite are not only affordable, but many of them are great performers! Used creatively and judiciously, signal processing can help you achieve professional-sounding results.

Without any kind of signal processing, the sound of everything we recorded would solely be dependent upon:

- The performance of our equipment.
- The sound of the instruments or voices.
- The acoustics of the room in which we recorded or monitored.

With accurate (and costly) equipment and beautiful room acoustics, some music is best recorded without any signal processing or special effects. Classical music, certainly, is one such music which most people prefer to record 'unadulterated.' When recording classical or other acoustic music, microphone placement determines how much of the room's natural reverberation is recorded, and is usually the primary means used to effect the 'color' of the recording. When the equipment is not the most accurate, however, and the room isn't Carnegie Hall, we need to consider ways in which we can alter the sound to achieve realistic results. And, more to the point, today's music frequently tries to achieve distinctly non-realistic results. In fact, ever since the introduction of the electric guitar, popular music has used electricity to *modify* sound, creating new sound colors, tones, and special effects—wailing guitar feedback, cavernous-sounding drums, and swirling synthesized sonic soundscapes are commonplace to our ears.

Types Of Signal Processors

There are several categories of signal processing equipment which create these and other special effects. The categories include:

- *Equalizers* (EQs), which are used to alter the frequency response—or tone—of a signal.
- *Delays*—used to simulate echoes, in addition to many special effects such as 'flanging,' 'phase shifting,' 'chorusing'—all of which use delay functions to create their swirling and shifting 'multi-dimensional' sounds.
- *Reverbs*, which can recreate the natural reverberation of acoustic spaces, such as rooms, concert halls, and even canyons, as well as perform special, 'unnatural' effects.
- *Dynamic controllers*, which control in different ways the amplitude levels of signals. Some dynamic controllers include *compressor/limiters*, *gates*, *expanders*, and other devices, and are used to help record with proper levels, reduce noise, and create special effects.
- *Miscellaneous processors*, which include unique devices such as *vocoders* (which allow an instrument to 'play' a voice), *exciters* (which enhance the apparent brightness of music in a way which changes with the music), and other devices.
- *Noise reduction*, which can be built into tape recorders or added as outboard gear, to reduce tape hiss and make for quieter recordings.

Virtually all mixers used for multi-track recording have some type of built-in equalization. A few mixers—particularly those oriented for live sound work—have built-in reverbs and delays, and some of the most expensive multi-track mixers have integral dynamic controllers.

Aside from EQ, however, most signal processors are in the form of *outboard* gear—that is, they are not built into the mixer, and are connected as outboard, auxiliary devices.

Digital Vs. Analog Processing

Pick up any current musicians' magazine, and you'll see advertisements for a host of signal processing products. Particularly in the field of reverbs and delays, *the* buzzword is *digital*. What does this mean when talking about an effect—does an effect have to be digital to be any good?

A true digital effect is one which processes all of the sound digitally, using a microprocessor (essentially, a computer). With digital effects, the incoming *analog* (non-digital) signal is converted to a series of binary computer-code numbers, called *bits*. By altering the structure of the computer codes, and reconverting the codes back to an analog signal, the sound can be manipulated in a number of different ways. This is an ideal method for reverbs, delays, and other devices to operate, since the codes can be stored for recall, and the digital process introduces no noise to the signal. (More about digital audio on page 124.)

In general, EQs and dynamic-controlling devices are analog, though this is starting to change. With analog gear, the signal is manipulated by altering the actual electronic signal (and not converting it to a digital code). In doing so, some of the analog device's own noise is introduced to the signal—this is the chief drawback of analog processing. Occasionally EQs and other devices are *digitally-controlled*—which does *not* necessarily mean that the sound entering and leaving the device is digitally processed. Digital control means

that different signal effect settings can be stored in the device's memory for later recall (just as with true digital effects). This storage/recall facility is ideal for changing the effect within a song, or for comparing different settings.

Just as the vacuum tube has all but succumbed to the transistor, though, analog gear is being gradually replaced by digital gear, not only for storage and recall, but theoretically better and quieter processing. As this happens, the smart shopper may be able to pick up some excellent though unfashionable equipment at bargain prices. Many analog devices—particularly EQs and dynamic processors—sound great, and introduce little discernible noise, distortion, or other sonic rubbish. In fact, there are engineers who prefer some of the older, 'warmer'-sounding (as in more harmonically distorted) equipment, and put it to creative use. With this kind of an approach, no equipment is ever truly obsolete!

Until multi-effect devices are available six at a time for a pittance (which may not be all that far away), most home recordists must prioritize outboard signal processing gear, starting with one or two pieces and adding slowly. In addition to built-in mixer EQ, a reverb unit is usually the first priority for most home recordists trying to achieve a quality sound, followed by a delay or compressor.

One promising trend is the incorporation of *several* effects into one box. While most of these units can only do one effect at a time, some effects boxes are *multi-tasking*, and can perform two or more effects—such as reverb, delay, EQ, and so on—at once!

Whether or not you purchase any outboard signal processors, chances are you already own several processors—the built-in equalizers in your mixer or recorder/mixer. Let's take a look at equalization. . . .

Equalization

All of us have used equalizers (EQs), if only the bass and treble controls of a home stereo. From those two controls, we know that *boosting* (raising the level of) the treble control makes music sound 'brighter.' We also know that *cutting* (lowering the level of) the treble tends to 'dull' or even 'muffle' the sound of music. Similarly, boosting or cutting the bass creates related changes in the tone of the lower frequencies.

In fact, what's going on when we fiddle with these controls is that we're increasing and decreasing the gain of specific frequency bands in the musical spectrum. By plugging an audio signal into an EQ, we can alter its frequency response—and by doing so, we're able to adjust the overall tone of music to suit our own taste.

Equalizers can be used on a single channel of audio, such as a bass guitar or vocal, or they can be used on an entire mix of music (as they are used in home stereos). By altering frequency response, EQs can be used to:

- Make a vocal track sound as if the singer is coming out of a telephone (by cutting all the low frequencies and boosting the upper-mid- frequencies).
- Reduce high-frequency tape noise or record surface noise (by cutting the appropriate high frequencies).
- Help a piano sound much more 'present' and 'unmuddy' (by cutting the appropriate mid-frequencies).
- Simulate stereo from a monaural instrument (as we'll learn on page 94).

These are just a few of the uses for equalizers. In their most basic applications, EQs can be used to compensate for microphones, speakers, and so on, in order to make an instrument or voice sound more natural—or at least more suited to our taste. In more advanced applications, EQs are used to create special effects such as the ones described above.

In these next few pages we're going to explore EQs and equalization—things may get a bit technical, but in order to truly understand the role of equalizers in recording, it's necessary to understand how they work, and what they do to your music.

Active Vs. Passive. Equalizers are designed with *active* or *passive* circuits. Active EQ circuits are actually very low-power amplifiers which are capable of boosting or cutting specified frequencies; passive EQ circuits are electronic filters designed solely to cut frequencies. For the most part, EQs designed for recording—including outboard and built-into-mixer devices—have active circuits. A common passive circuit is the tone control found on most electric guitars. By turning this control counterclockwise, it 'filters out' high frequencies, making the guitar sound more 'muffled.' Since passive EQ circuits don't use any amplifier stages, they don't require any AC or DC power to operate. Filters are still found in many synthesizers, and filter-only EQs are sometimes used to equalize the frequency response of monitor speakers.

Bands. A simple bass and treble control—such as those found on many recorder/mixers—is known as a *2-band* EQ. The number of bands that an equalizer has refers to the number of segments of the audio spectrum (20 Hz to 20 kHz, approximately) that the EQ is able to effect individually. So a 2-band EQ divides the audio spectrum into just two bands, a 10-band EQ divides it into 10, and so on.

A typical 2-band bass and treble EQ doesn't offer the ability to control sound in a refined manner, since adjusting one frequency tends to affect a broad band of frequencies. Think of trying to use a single treble control on a home stereo to reduce tape hiss or record surface noise: When you do so you also cut out a lot of music in the high-end! Trying to reduce a rumbling noise with just a bass control also cuts out musical information. Consequently, a 2-band EQ can be used to adjust the overall tone of an instrument or a piece of music, but not specific frequencies. We'll learn more about bands a bit further ahead, under "Graphic EQs."

EQ Response Characteristics. Let's consider a simple 3-band EQ, with a bass, a 'midrange,' and a treble control. Figure 8.1 shows us a graph of such an EQ's *response curves*—the lines which represent the range of frequencies which are affected when the EQ's bands are at maximum cut or boost.

What this graph shows us is the range of effect of the three bands, the bass (low frequency) band on the left, the midrange in the middle, and the treble (high frequency) band on the right. The half of each band above the "0 dB" mark shows the maximum range of boost for each band; the half of each band below the "0 dB" mark shows the maximum range of cut.

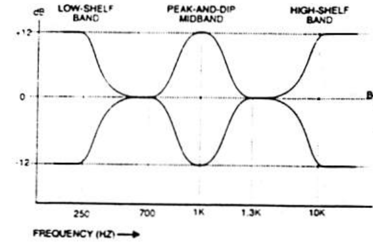


Figure 8.1 Response curves of a typical 3-band equalizer, with shelving low and high bands, and a peak-and-dip mid-band.

Notice how the bass and treble bands look different from the midrange band? As with many 2- and 3-band EQs, these low and high frequency bands have a *shelving* response: A *high-shelf* band is one which introduces boost or cut at a *shelf frequency* (also called *hinge frequency*), and then 'flattens out' to boost or cut everything above that frequency to the same degree. Similarly, a *low-shelf* band can boost or cut a shelf frequency, and then flatten out to boost or cut everything *below* that frequency to the same degree.

The midrange band in Figure 8.1 on the other hand, represents a *peak-and-dip* response. Peak-and-dip EQs are exactly as they sound to be: They 'peak' when boosted, and 'dip' when cut, at a *center frequency*—and have a lesser effect on frequencies lower or higher than the center frequency. The center frequency of the midrange band pictured is 1 kHz. As we'll learn about starting on page 45 some EQs allow you to change the center frequencies of one or more bands. For now, though, we're dealing with *fixed-frequency* EQ bands.

Musically, shelving and peak-and-dip bands are quite different from each other. Referring once again to Figure 8.1 take a look at what happens when the high frequency band is cut all the way: All frequencies above 10 kHz or so are cut—and that's why turning down a treble control to reduce hiss at 16 kHz also reduces response around 10 kHz, which can reduce the top end of cymbals and other instruments. Peak-and-dip bands are much more useful for controlling specific instruments or frequencies. A peak-and-dip band centered at 3 kHz, for example, would be able to 'brighten' the sound of a snare drum without boosting hiss.

Shelving EQs do have an important role to play, however, and if used carefully can be quite effective. If a mix is too bright, a high-shelving EQ can be used to lower the overall treble content—it's a lot more straightforward than tying up lots of graphic bands. For these and other reasons, shelving EQs are frequently found not only on recorder/mixers, but also as the high- and low-bands on numerous mixers.

Figure 8.1 also shows us that the amount of boost or cut available from each band is measured in terms of decibels—the typical range of boost and cut is 12 or 15 dB in each direction. The range of control for the bass

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band in Figure 8.1 can be expressed as *250 Hz (shelving response), ±12 dB*. Remember that every 10 dB is an approximate doubling in apparent volume—so that much of the range of a bass guitar (which is 40 to 440 Hz) will sound louder if we boost a bass control hinged at 250 Hz by 10 dB.

We know now that high-shelf bands affect frequencies above their hinge frequencies, and low-shelf bands affect below their hinges. Peak-and-dip bands, however, affect a limited range of frequencies above or below their center frequencies. For example, we don't boost 1 kHz exclusively when we increase the midrange control. In fact, with the control shown we boost frequencies all the way from 700 to 1300 Hz, though the greatest degree of effect occurs at the center frequency. This range of effect is known as *bandwidth* (page 46). Not all EQs have the same bandwidth for each band, and some EQs allow you to adjust the bandwidth, as we'll learn more about shortly.

* * * *

So far, there are several things we've learned about EQs:

- An EQ can be designed with either active or passive circuits. Active circuits are most common.
- EQs have bands, and each band covers a specified frequency range.
- Bands have shelving or peak-and-dip response curves. Shelving is most common with the highest and lowest bands of 2- or 3-band EQs; those bands are known respectively as high-shelf and low-shelf bands.
- Shelving bands have shelf, or hinge, frequencies at above or below which the EQ can boost or cut.
- Peak-and-dip bands have a center frequency at which the maximum boost or cut takes place.
- When the center or hinge frequency is non-adjustable, the band is a fixed-frequency band.
- Each band can be boosted or cut by a specific number of dBs.
- The frequencies which are affected by boosting or cutting a peak-and-dip EQ band represent its bandwidth.

We've been describing simple 2-and 3-band fixed-frequency EQs up to this point. There are several other types of EQs commonly used in recording, including:

- *Graphic* EQs.
- *Sweep-frequency*, or *semi-parametric* EQs.
- *Parametric* EQs.

Graphic EQs. Graphic EQs are perhaps the most popular type of equalizer, since they are quite easy to use and can be relatively affordable. They are usually single- or dual-channel outboard units, with anywhere from five to 62 fixed-frequency bands per channel—though the most common are 10-, 15-, 27-, and 31-band graphics. Almost all graphic EQs found in the recording world are outboard units.

Most graphic EQs use a series of linear sliders—one for each frequency band of the audio spectrum—to adjust the sound. The sliders are arranged with the lowest frequencies on the left, and the highest frequencies on the right. When adjusted, the various physical settings of these sliders provide a rough graphic representation of which frequencies the EQ is altering, and to what degree. Consequently, they allow you to change signals in a visually-predictive way.

For example, let's say we were listening to a sustained synthesizer chord through a mixer, and the music had a fairly flat frequency response. Now let's say we had a 10-band graphic EQ, and connected it in between the output of the synthesizer and the input to the mixer. Altering the controls of the graphic EQ would produce predictable responses, as seen in Figure 8.2.

Figure 8.2 also shows the response curve a 31-band graphic EQ.

Notice that as the number of bands increases, the bandwidth—the plus or minus frequency range of each center frequency—decreases. That's to say, the more bands, the finer the *resolution*, or degree of control. With a 10-band EQ, for example, high-frequency noise around 16 kHz can be controlled with one band of EQ, without affecting musical material around 10 kHz. We couldn't do this with a 2-band EQ because it has less resolution—trying to reduce 16 kHz noise would reduce 10 kHz music content. But with a 31-band EQ, for example, we'd be able to cut down the level of a 16 kHz clicking noise while boosting the uppermost harmonics of a cymbal at 14 kHz—something we couldn't do with a 10-band graphic.

A 10-band EQ is also known as an octave EQ, because its band controls have center frequencies which increase by one octave at a time (a doubling or halving of a frequency is a change of one musical octave). A typical 10-band EQ has its bands centered on *31, 62, 125, 250, 500, 1 k, 2 k, 4 k, 8 k, and 16 k* Hertz. In a similar manner, a 31-band EQ is known as a *one-third* octave EQ, since each center frequency is one-third of an octave apart, and a 15-band EQ is known as a two-thirds octave EQ. But what do all of these numbers have to do with *music*?

The answer lies in remembering that music—or more generally, audio—covers an approximate frequency range of 20 Hz to 20 kHz. A single fundamental note of an instrument may have harmonics which go way beyond the fundamental frequency (as we learned on page 4. Being able to cut or boost not only the fundamental but also the harmonics is what allows us to change the *tone quality* of an audio signal. Because a graphic EQ's bands cover most of the audio spectrum, many people use them to shape the overall tone of an instrument or voice. In this manner, many of the bands may stay set at '0 db'—where they aren't affecting the frequencies—while other bands are boost or cut as necessary.

One-third octave graphic EQs are also popular in live sound reinforcement, where feedback can be squelched quickly by cutting the offending frequency. Graphic EQs are often employed in conjunction with a real-time analyzer—as is described ahead on page 74—to adjust the frequency re-

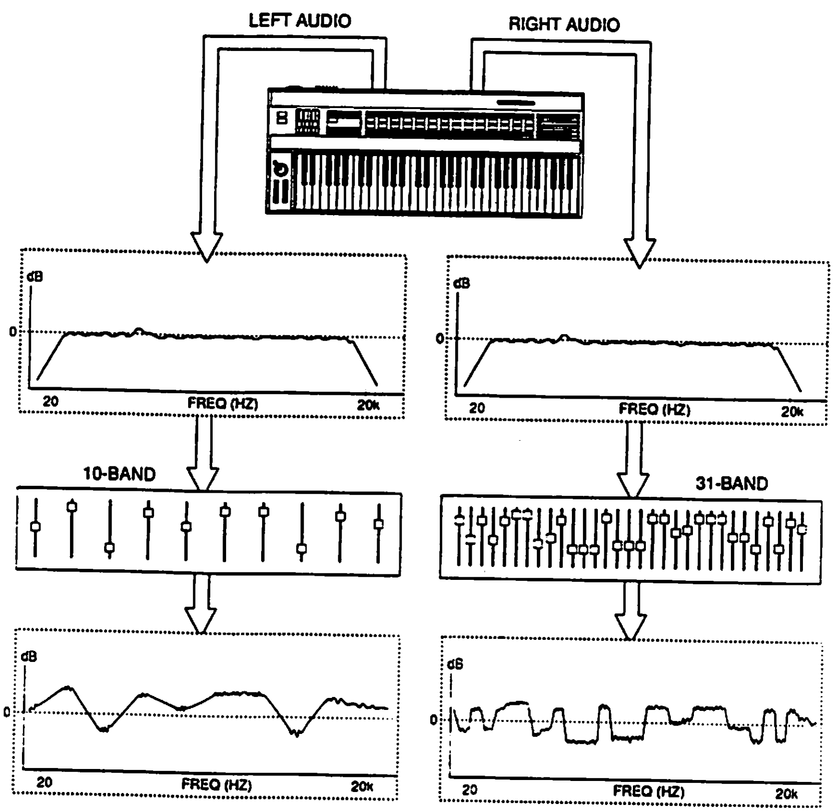


Figure 8.2 Graphic EQs offer a graphic representation of how they affect sound. As we can see, more bands offer greater resolution, and offer greater control over how sound is shaped. (One-third octave EQs are available with anywhere from 27 to 31 bands.)

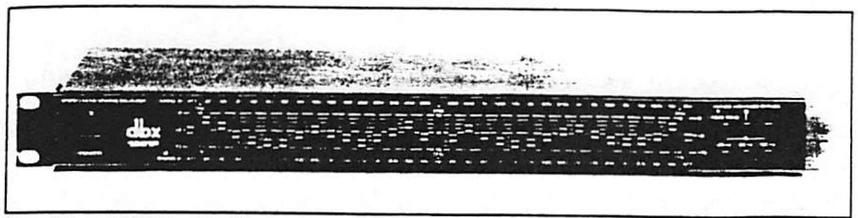


Figure 8.3 dbx's 1531P can function as either a dual channel 15-band EQ, or as a single channel 30-band EQ.

sponse of loudspeakers, so that they perform as accurately as possible in a given room.

There are, by the way, some graphic EQs which aren't really graphics: These fixed-frequency EQs, such as the one-third octave White 802, use rotary instead of linear controls for each band, and can be treated in every way as graphics (except that they don't provide a 'graphic' picture of their effect).

Sweep-Frequency EQs. So far, we've been talking about fixed-frequency EQs, where each band is centered at an unchangeable frequency. The next step up in the hierarchy of EQs is one which allows you to select *which* frequency can be cut or boosted for one or more bands. This is known as a *sweep-frequency* EQ. It's also called a *semi-parametric*, or just plain *sweep* EQ.

There are a few 10-band graphic sweep-frequency EQs on the market, such as the Orban 674A, and they're wonderful, since they provide a graphic visual reference along with a secondary frequency control for each band. For the most part, however, sweep EQs are 2- to 4-band equalizers—built into mixers and recorder/mixers, or outboard—and allow you to adjust the center frequency of one or more of the bands.

A common arrangement with many recorder/mixers is a sweepable 2-band EQ on each input. Even though these are only 2-band EQs, the fact that you can select which center frequency (or hinge frequency, if it's a 2-band sweepable *shelving* EQ) you're cutting or boosting greatly enhances their flexibility.

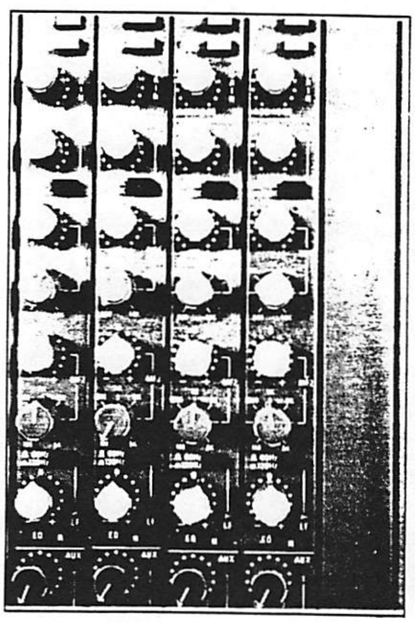


Figure 8.4 The EQ section of a TAC Scorpion mixing console.

Figure 8.4 shows a closeup of the TAC Scorpion's 4-band equalizer. Note that the

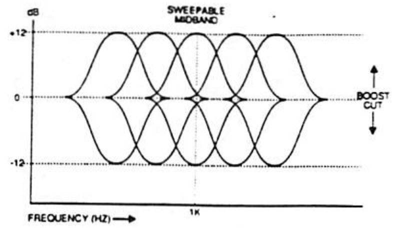


Figure 8.5 A 'sweep mid' control offers a range of cut or boost.

high- and low-frequency bands (which are shelving) each have selectable hinge frequencies. The two mid-bands (both peak-and-dip) can be cut or boosted over a very wide range of frequencies—the upper mid-band can sweep from 500 Hz to 18 kHz, the lower mid from 100 Hz to 5 kHz. This design is known, in insider's lingo, as a '4-band, sweepable mids' EQ.

A sweep EQ can be really handy; the ability to choose *which* frequencies you're

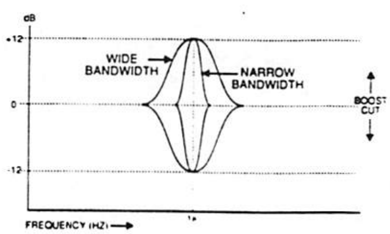


Figure 8.6 Comparative bandwidths.

equalizing provides some pretty powerful tone control. "Basic EQ Tips," on page 47, discusses some things to bear in mind when using EQs, and with a sweep EQ you can take advantage of each of those suggestions.

Parametric EQ. Whereas graphic EQs are useful for general tone modifications and are easy to use, and sweep EQs are common and useful, a true parametric EQ offers unparalleled tone control. As we've learned, graphic EQs allow you to boost or cut at fixed frequencies and bandwidths: sweep EQs allow you to choose which frequency is being cut or boosted, giving you greater control and flexibility. Parametric EQs, on the other hand, have everything going for them. Each parametric band has three controls which allow you to:

- Select a center frequency for each band.
- Cut and boost on each band.
- Adjust the bandwidth of each band.

Look back at Figure 8.1—and take note of the midrange frequency control's EQ curve. If we could control the bandwidth of just that frequency and band, we could have curves which look like Figure 8.6.

This combination of frequency and bandwidth control allows music to be tailored with a wonderful degree of precision. Unfortunately, all of this control can be difficult to use effectively, and offers a real challenge to the user. For example, when the bandwidth is set to a minimum, it may be impossible to hear the EQ do anything, especially if there's no musical content at the frequency selected!

Specialized Filters. There are four other types of equalization we haven't discussed. All of these are known as filters, since they cut frequencies, though they can be active or passive (again, passive EQs don't require any power to operate).

The most common of these are *high-pass* and *low-pass* filters. Many people are familiar with these controls from their home stereos: The switch marked *subsonic* (or sometimes *low filter*)—which removes low frequency rumble—is actually a high-pass filter. The the switch marked *high filter*—which removes high frequency hiss and the like—is actually a low-pass filter. Confusing but true! In any event, high-pass filters are quite common, as a single switch, in some better consoles. Low-pass, as well as high-pass filters, are frequently found as switches or variable controls in many top consoles as well as outboard EQs.

Here's what they do: High-pass filters are designed to cut out bass frequencies, and let everything *higher* than a certain frequency be heard. They're usually fixed between 60 and 120 Hz or so, though sometimes are variable on better consoles and outboard EQs. High-pass filters have a really steep roll-off (usually 24 dB/octave), designed to minimize any sound below the pass frequency. This type of filter is useful for eliminating all types of unwanted low-frequency noise and rumble, everything from trucks rolling by on the street to a singer's footsteps, which can be picked up by a microphone (via a floor stand). With certain instruments—such as a bass guitar—a high-pass filter can make the sound pretty thin, by

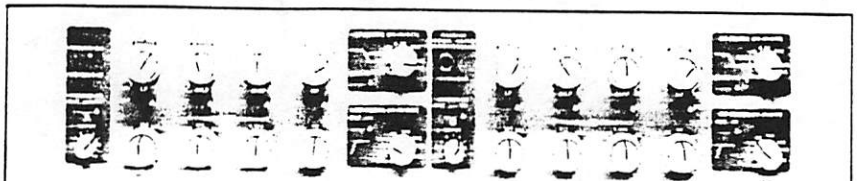


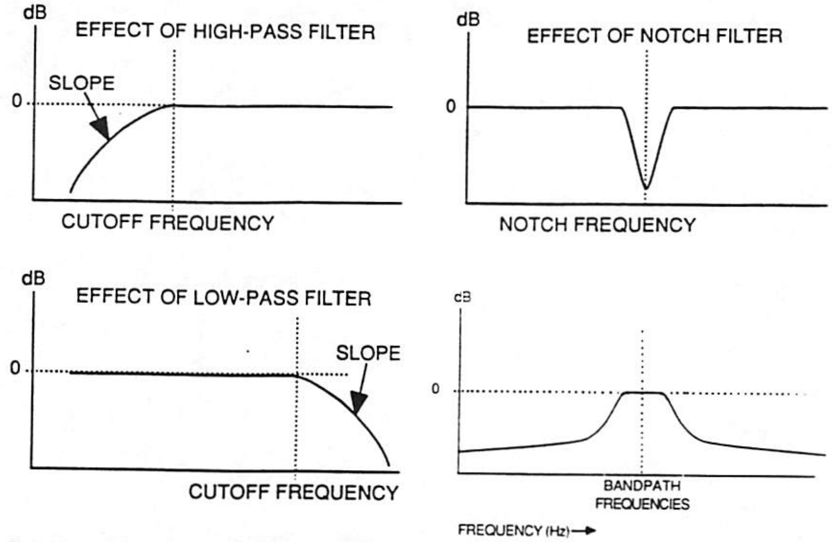
Figure 8.7 This parametric EQ by Orban can function either as a dual-channel 4-band, or single-channel 8-band EQ. It also includes low- and high-pass filters and a fine-tune 'vernier' control for the frequency of each band.

removing all the essential low-end information. Some engineers, however, prefer to engage them all the time (providing they're not adversely affecting the signal).

Low-pass filters let everything *lower* than a selected frequency pass through and be heard. They are most useful for getting rid of excessive high-end noise above 12 kHz or so. A low-pass filter can make things sound dull, though, especially cymbals.

You may run into *notch filters*, which are interesting and useful tools in the studio as well as for live sound. Simply, a notch filter allows you to 'zero in' on a particular frequency and effectively remove it from the signal. We've all been at concerts (or participated in them!) when the grit-your-teeth sound of feedback comes screaming through the speakers, only to be picked up by the microphones again and re-amplified. A notch filter can be used in such a circumstance to find the offending frequency of the feedback and cut it out.

Essentially, a notch filter band is centered at a particular frequency (usually variable) and is able to cut 30 dB or more (up to - ∞) at that frequency, by having a curve with super-steep slopes and a very narrow bandwidth. In a recording situation, a notch filter can be used to help reduce the squeak from a piano pedal, a 'boomy' frequency from a kick drum, as well as the North American 60 Hz (or 50 Hz elsewhere) hum which can find its way onto a tape of otherwise great music (more about this hum starting on page 81). Notch filters are rarely found in consoles: If not a separate piece of outboard gear, they are often found included in a parametric equalizer.



Figures 8.8a, 8.8b, 8.8c, 8.8d Various filter responses.

Finally, a *band-pass* filter is more or less the opposite of a notch filter, in that it singles out a particular frequency and attenuates all other frequencies high and low of the chosen one. These are rare to find these days (outside of synthesizers), as their recording applications are limited.

Understanding Bandwidth

As we know, 'bandwidth' refers to the range of frequencies which are effected by an EQ band. In order to use your EQ to its full sonic potential, it's really important to understand your EQ's bandwidth—whether you have simple 2- or 3-band, graphic, sweep, or parametric EQs.

Parametric EQs allow you to vary the bandwidth. Some of the better mixers may have 'quasi-parametric' EQ bands which offer a switch to choose a wide or narrow bandwidth for one or more EQ bands. All other EQ bands are fixed-bandwidth. The term Q, for "quality factor," is used to express an EQ's bandwidth. Many people use 'Q' and 'bandwidth' interchangeably, though they're not necessarily the same term, depending upon who you talk to. Bandwidth is typically expressed in octaves, so that a one-third octave EQ band has a bandwidth of 0.33 octaves. The higher that figure, the wider the bandwidth.

Instead of referring to the entire bandwidth range, Q is a figure which represents a ratio of the center frequency divided by the '3 dB down' frequencies on either side of the center frequency. It's graph time again:

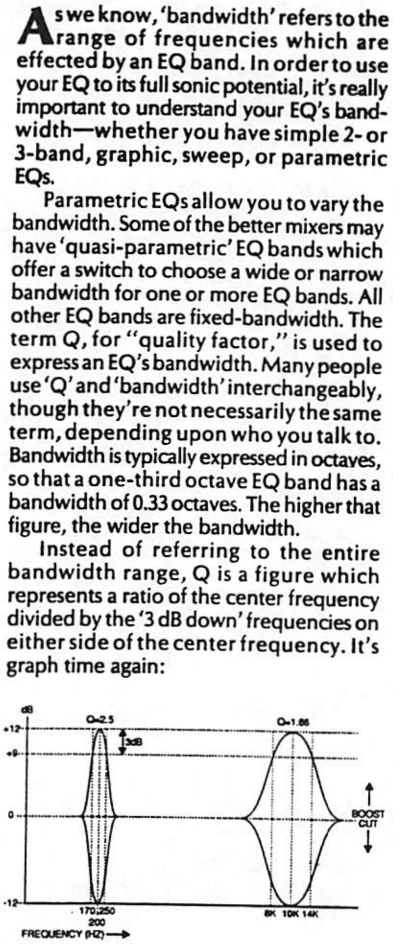


Figure 8.9 'Q.'

In Figure 8.9 we can see two bands of peak-and-dip EQ. The low-frequency band (on the left) has a very *narrow* bandwidth, and the high-frequency band has a very *wide* bandwidth. Take a look at the low-frequency band, and find the frequencies on either side of the center frequency (200 Hz) which are 3 dB down in level from 200 Hz. They are 250 Hz and 170 Hz. Now we can use a formula to calculate the Q:

$$Q = \frac{\text{CENTER FREQUENCY}}{\{\text{HI FREQ — LO FREQ}\}}$$

[Where "Hi" and "Lo" frequencies are measured at -3 dB down from the center frequency.]

or in other words:

$$Q = \frac{200}{\{250-170\}}$$

$$= 2.5$$

So in this case, the Q is 2.5. Try calculating the Q for the high-frequency band. Using the same formula, you should come up with a Q of 1.66. So now we've figured out that—unlike octave representations of bandwidth—the higher the Q number, the narrower the bandwidth.

Parametrics offer variable bandwidths, and the amount of Q that one should select will, of course, depend upon the music one wishes to EQ. Similarly, if you don't own a parametric, knowing the approximate bandwidth of the frequencies you wish to adjust will help you immensely.

For example, let's say we wanted to brighten the sound of an acoustic guitar. A broad boost of its higher harmonics, which might be from 3 to 8 kHz, would be appropriate. With a graphic EQ, this could either be done by boosting bands from 3 to 8 kHz by 1 to 3 dB. With a parametric EQ, you might achieve the same by centering a band at 5500 Hz, setting the band for maximum bandwidth, and boosting appropriately. If, on the other hand, there was a particular frequency which would sound great being boosted (or an offending frequency which begs to be cut), a single graphic band or a narrow Q setting on a parametric would be the appropriate tool to use.

The term *rolloff* (or *slope*), by the way—as indicated in Figure 8.8a—refers to the rate at which the EQ either gains or loses its effect on either side of the center frequency. It's expressed in terms of *dB/octave* (decibels per octave), and helps determine the bandwidth.

When bandwidth is not adjustable, most EQs are designed to be 'musical,' so that their bands cover the frequency ranges which the design engineers consider most effective. Whether fixed or adjustable, however, EQs should have a 'constant-Q' design—so that once set, each band's bandwidth stays the same regardless of the amount of cut or boost. Without this type of design, you'll be affecting a different range of frequencies depending upon how much EQ-action each band is getting.

The most effective way to use a notch or band-pass filter is to set it to maximum cut or boost, then adjust the frequency control. If it's a notch filter and you're using it to eliminate feedback, the feedback should disappear as soon as you dial in the offending frequency.

EQs And Amplitude. One thing to keep in mind when you use EQs is that they can alter the overall level of any signal passing through them.

For example, when you boost any band of EQ, you increase the total level of the track being processed. When a band of EQ is cut, that track's level is also cut. This is perhaps most important to remember with graphic EQs: If every band of a 31-band EQ is boosted 6 dB, then (in theory) an instrument passing through that EQ will sound the same as if it were passing through the EQ with every band set to '0'—except that it will be louder, and there will be more noise in the signal.

The noise factor is an important key: Remember that EQs use amplifiers to boost signals, and even the best of amplifiers introduce noise. You can hear your equalizer's inherent 'self' noise by just listening to it *without* any signal passing through. As certain bands are boosted or cut, you'll be able to hear noise being introduced (or cut) from the system.

Take a look at Figure 8.10. What we have here are two EQs, each with the same EQ settings. The EQ on the bottom is introducing a lot of extra noise, however, since so much extra boosting of the signal is happening—not only are musical frequencies being boosted, but so is noise. The general rule of thumb for using EQs is that usually all the settings should average around zero, as we can see with the EQ on the left.

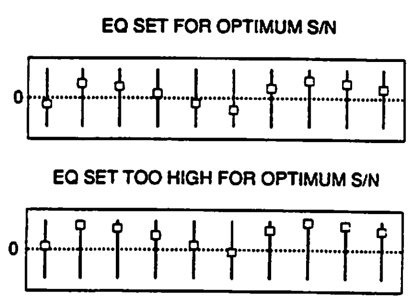


Figure 8.10

Basic EQ Tips

Equalization is not, in all cases, the best way to control the tone quality of an instrument or voice. As we've learned, active EQs can introduce noise and distortion to your signal. They can also cause subtle phase shifts, which can further degrade your signal. And some of the cheaper models use filters which can actually cause a 'ringing' sound!

For these and other reasons, some engineers prefer to use EQ as a last resort for modifying a tone, and would prefer to reprogram a synthesizer or change a mike's placement before using EQ. Related to this, a general rule of thumb to apply to equalization is that *less is more*. This is not only to minimize the sonic problems just mentioned: Too much EQ can create an imbalanced sound. Subtle use of EQ is often more musically effective than huge 10 dB cuts and boosts all over the spectrum. This minimalist approach can be applied to much of the recording process, in fact, through signal processors to the actual arrangement and production of the music.

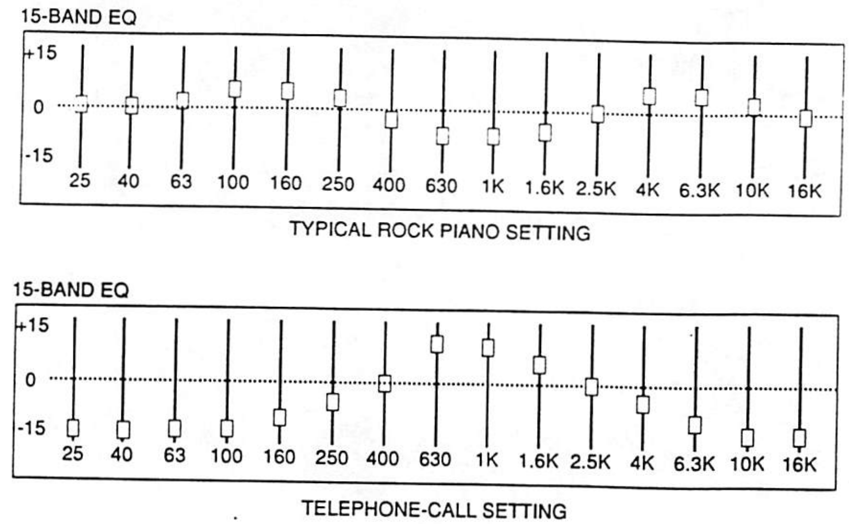
Still, equalization remains a powerful tool, and its benefits usually outweigh its drawbacks—particularly if you understand your EQs. Your ears are the best tools for *learning* about EQ, and how to create effective sonic tones. Because of this, the most important tip about EQs (as well as all signal processing gear) is to *experiment!* If you own a graphic EQ, what happens when you boost just one band and cut all the others? If your mixer or recorder/mixer has sweepable bands of EQ, try playing a steady chord (or drum machine pattern, or whatever) through a channel, and listen to the effect of sweeping to different frequencies and cutting or boosting.

As you experiment and listen closely and critically, you'll improve your EQ technique. Listen not only to the effect you are having on the music, but listen for the 'side effects.' If you're equalizing a bass guitar, does boosting around 10 kHz do anything, or does it just add noise? Similarly, does cutting around 500 Hz detract from the 'bottom end' of the bass? Does that depend upon which notes are being played? Remember, most instruments cover a broad range, and just the fundamental notes may cross over several EQ bands.

By using your ears, you'll not only learn which bands and degree of effect are useful to achieve the sound you want with the instruments you're recording, but you'll also learn what's not so useful. Here are some general tips which may help that process:

- **Beware of 'frequency masking.'** An instrument which has been EQ'd to sound great on its own may not always sound so great when it's 'in the mix' with several other instruments. For example, a warm, muted guitar with very little high frequency information might sound wonderful on its own, but may be lost when played back with keyboards and lots of background vocals. This is because our ears make a kind of composite signal of all sounds happening at once. When several instruments emphasize similar frequencies, such as a muted guitar, keyboards, and vocals, those frequencies accumulate, and can become overbearing or cause one instrument to 'mask' another. Be prepared to re-equalize tracks when mixdown time arrives, or better yet, think ahead and try to 'visualize' the frequencies you'll have to contend with when all the tracks are done. This kind of 'frequency inventory' can help you plan appropriate EQ settings.
- **The low end.** Very few loudspeakers are capable of accurate bass reproduction below 40 Hz. So go easy down there—unless your speakers have phenomenal bass response, it will be difficult to equalize, with any accuracy, frequencies much lower than 40 Hz. Those of us with smaller speakers especially may have a tendency to boost bass frequencies—but what winds up sounding good on our little speakers may sound terrible on a larger pair. The useful low frequencies to pay attention to are from 40 to 250 Hz, where you can find kick drums, keyboards, some vocals, and even the lower registers of a guitar. While some degree of boost can add 'body' to kick drums and male vocals, guitars and other instruments can sound too muddy or boomy if these frequencies are boosted too much.
- **The midrange.** The midrange frequencies (250 Hz to 2 kHz) seem to accumulate an excess amount of information sooner than other frequencies. It's helpful to keep this in mind when building tracks. A piano or guitar, for example, can become quite muddy-sounding if EQ'd with too much midrange. Often a slight cut between 500 and 1200 Hz will help. Don't go overboard, though—it's easy to create a mix that's all 'sizzle' and 'thump,' with too little midrange information, particularly when there are mostly acoustic instruments. The all-important human voice can be found in the midrange, and it's important not to make it too thin by equalizing out the warmth and humanity.
- **The upper-midrange.** Many engineers and producers use the term 'definition'—which can mean a lot of things, but usually refers to the intelligibility, or 'cut' of a particular instrument or voice. The upper-midrange and lower-high-frequency areas of 2 to 6 kHz contain many of the harmonics which, when carefully EQ'd, can improve an instrument's definition. For instance, if you wish to bring the 'snap' of a 'slapped' bass, try boosting around 3 kHz. Many other string instruments in particular, including violins, mandolins, guitars, and the like, have harmonics in this range. Listen carefully—too much boost can make things sound harsh, and too much cut can make a mix sound dull and lifeless.
- **The high end.** The highest frequencies from 6 to 15 kHz (and higher) contain mostly harmonic information, though there are some fundamentals of cymbals, synthesizers, and other instruments residing in this 'top end' of the audio spectrum. Listen closely to each track. Does cutting the high end slightly reduce noise and *not* affect the track? Or does a slight boost add 'life' and 'sparkle' to the sound? Follow your ear, but be careful boosting up here—tape hiss and other noise can benefit more than the music.
- **How's your reference?** Your ears are only as good as the monitor chain. As we'll learn on page 69 it's critical that you use speakers which you can *trust* to deliver a reasonably accurate picture of the sound you're creating. Whenever possible, give yourself 'second opinions' by listening to your music through a car stereo, a portable stereo, or even a friend's stereo. Even in the best studios, a great studio mix can fall apart in a 'real-life' situation.

One common tendency when using EQs in music is to cut the middle frequencies—say those from 500 to 2000 Hz. The audible result with pianos, guitars, and other *mid-range* instruments is that the signal becomes 'cleaner,' with less mid-range 'presence.' For example, Figure 8.11a shows a typical 15-band graphic setting that might work well with a piano:



Figures 8.11a, 8.11b Equalizers can be used for a variety of purposes, from tailoring an instrument's sound, to creating a special effect. Typically, when pianos are heard 'in the mix' with other instruments, they tend to lose definition, since many of their important frequencies are masked by other tracks. Cutting the mids and boosting the lows and upper-mids tends to help many pianos 'cut through' the mix. Similarly, the 'peaky' sound so commonly associated with telephones is in part related to the fact that telephone lines boost mid frequencies—those frequencies where the voice is—in order to provide greater definition. The setting shown mimics that response.

Using the 'graphic' analogy, we can see that the mid-frequencies of 630 and 1000 Hz have been cut by about 6 dB, and some lower and higher frequencies have been boosted. Try to visualize how a piano might sound being equalized in this manner. It should sound brighter and bassier, because of the frequencies we're boosting. Similarly, many of the mid-range tones would be played down, perhaps making the piano sound a bit 'thinner.'

To many people, such a setting makes the piano more intelligible, and gives it more 'cut,' especially when it's recorded with several other instruments. That's because most recordings contain a lot of mid-range frequencies, from vocals to keyboards and guitars—sometimes the recording can benefit by reducing some of these frequencies. For many engineers, however, a much better alternative to having to EQ a piano in this way is to mike it in such a manner that a similar frequency response is achieved.

To demonstrate another application of a graphic, take a look at Figure 8.11b.

Can you visualize what your voice would sound like coming through an EQ with this setting? As we can see, almost all the low end is cut, as is a lot of the upper-high end. In addition, the mid-frequencies are radically boosted. If you have a EQ graphic, try this setting with a vocal—it should sound as if you're singing through a telephone!

No matter what your EQ—a 2-band built into your ministudio, or an elaborate outboard parametric—listen and experiment. Compare other recordings, professional or amateur, with your own. Are the vocals equalized in the same way? Can you tell which frequencies are favored by different producers and engineers with different instruments? Do EQ settings stay the same with the same instruments from song to song, or do they change? Do settings even remain the same within one song?

The answers to these and other questions will describe your recording's 'sound color'—a critical element in your overall artistic statement.

Acoustics And Spatial Effects

Many of the most useful signal processing devices on the market are *spatial effects.* Not to be confused with *special effects* (though they can perform them), spatial effects are devices that alter the perceived acoustic space of a sound. These devices can reproduce natural effects, such as *echo, delay,* or *reverberation,* or they can create true 'special' effects, such as *flanging* or *chorusing.*

There are quite a few varieties of spatial effects devices, including:

- Delays (including tape, analog, and digital delays), which are used to create single or repeating echoes, as well as flanging and chorusing effects.
- Reverbs, which simulate acoustic spaces such as rooms, halls, and even caverns.
- Phase shifters, as well as independent flangers, and chorus devices, all designed to create specialized, swirling, '3-dimensional' sonic effects.

One common thread linking all of these devices is their ability to *manipulate sound over time.* Sound and time are inextricably linked in our world—to appreciate this, an understanding of the fundamentals of acoustics is necessary, and will help us get the most out of our delays, reverbs, and other spatial effects.

Echoes, Reflections, And Time. *The Grand Canyon. The Swiss Alps. The Taj Mahal.* If we paint a sonic picture of these places in our minds, we can imagine repetitive echoes and cavernous sounds, expanding a casual yodel into a magical chorus of yodels. These places can do their sonic tricks because the physical shape of their environments cause sound to *reflect* back to its source. The time that elapses between the original sound and the returning reflected sounds is known as the *delay.* Our ears are remarkable instruments, and as we'll learn, we're capable of hearing astonishingly small delays in sound. Let's take a look at 'real-life' echo and delay, and while we're at it, we'll get in a short lesson on acoustics.

An echo is a repeat of a sound. There are two types of real-life echoes:

- A *single-repeat* echo, which is often called a *slapback echo* by the engineering cogniscenti.
- A *multiple-repeat* echo, also known as a *multiple delay.*

A single echo occurs when a sound is reflected back to its source just once. If you'd like to try some acoustic experiments, and don't live in the Grand Canyon, try this: Find a flat, smooth, hard-surfaced wall, face it head on from a distance of about 10 to 20 meters (about 30 to 70 feet), and clap your hands sharply, just once. If the wall is reflective enough (that is, smooth and hard enough—concrete is ideal), and you've faced it head on (so that the sound of your hand clap will reflect and return to you), you should hear an echo!

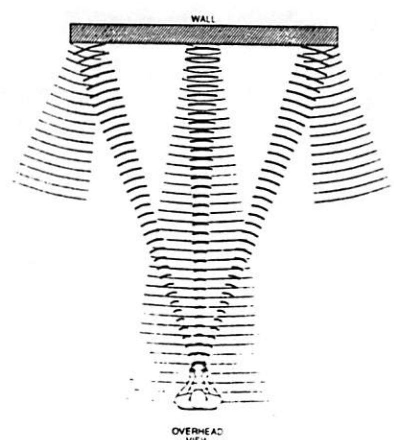


Figure 8.12 Cheap fun with walls.

With a single wall, we'll only hear one echo. After the delayed sound returns to our ears, it carries on past us, and we can no longer hear it.

Next, find two parallel and reflective walls, at least 10 meters apart—ensure that the walls are truly parallel. For now, try and find a space without any connection between the two walls—a path between two concrete buildings is a good choice. Position yourself equidistant between the two walls, let out a mighty clap, and listen to the effect.

If your site selection has been careful, you should hear an *amazing* ricochet sound, as your clap hits a wall, flies back past your ears, hits the opposite wall, flies past your ears, and so on!—until it *decays* to the point at which you can't hear it. With this type of continuous reflection between parallel surfaces *standing waves* can also occur. When a sound wave keeps reflecting and colliding back upon itself, the resulting 'standing

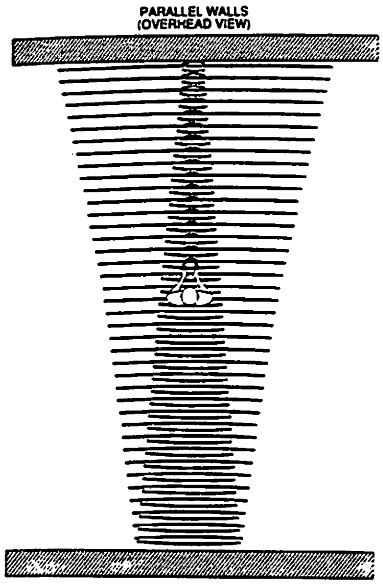


Figure 8.13 Parallel walls can yield repeating reflections, and 'standing' waves.'

wave' is similar to what happens when two water waves collide (see Figure 8.13), with some frequencies cancelling and others adding. These little tests can be as wonderful to hear as the latest studio effect, and they're free—you may even find a great sounding place to record.

Why smooth and hard walls? If a reflective surface is smooth and hard, it will reflect sound in much the same way that a mirror will reflect light. The degree of smoothness describes to what degree the surface *diffuses* the sound, and the hardness describes how *absorbant* the surface is. Many musicians know that drapes, carpeting, and other soft surfaces can be used to help control sound. Such soft surfaces help to control reflections by absorbing higher frequencies, and diffusing reflections.

The nice thing about these tests is that it's easy to visualize what's happening to the sound source. Some experimentation can yield some pretty interesting results. For example, with the single wall, what happens when you move away from the wall? Up to what distance can you still hear an echo? And what happens when you approach that single wall? How close can you get before you know longer hear a distinct echo?

As you may know, sound travels at about 335 meters per second (760 miles per hour) at sea level. It's this relatively slow speed that lets us hear distinct echoes. If you perform the single wall test, you should find that an echo becomes difficult to discern once you get within 9 meters (30 feet) or so. That's because at 9 meters (a round trip of 18 meters), sound takes about 50 milliseconds (abbreviated *ms*, for thousandths of a second) to reach the wall and return to our ears. Sounds which are much less than 50 ms apart from each other in time are no longer discernible as distinct, individual sounds. Because of this, once we hear the initial, unreflected sound of the clap, any echo must take longer than 50 ms to reach us in order to be heard as a separate sound.

When evaluating distance and time, one rule of thumb is that sound travels at a speed of one foot per millisecond; thus, to hear a sound that takes 80 ms to return to its source, position yourself about 40 feet away from the reflective surface.

Reverberation. Our two-wall example above showed us how reflected sounds can be re-reflected. In a case of two parallel walls, the effect is most startling since the reflections are so 'perfect'—that is, smooth and hard parallel walls cause predictable, repeating reflections. In most cases of real-life echo, however, we hear less-than-perfect, *random* echoes, which occur as sounds hit uneven surfaces and reflect in all sorts of directions. Let's take the Taj Mahal, for example.

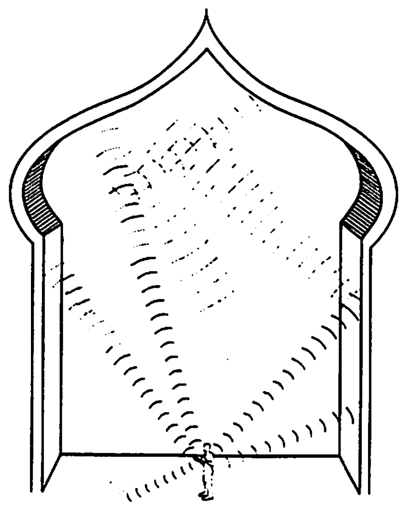


Figure 8.14 Not quite the Taj Mahal, but you get the idea.

As we can see in Figure 8.14, a hand clap in the Taj Mahal is reflected in every which way, with some reflections returning quickly to the source, and others taking much longer. In fact, a hand clap in the center 'sweet spot' of the Taj will echo and *reverberate* for about 30 seconds at night, and 20 seconds in the day (the higher humidity of the day causes quicker decay of the sound).

Notice the word 'reverberate': Repetitive echoes less than 50 ms apart are known as reverberation, or just plain *reverb*.

In a typical concert hall setting, a person in the audience is able to hear many different types of sounds, some directly from the stage, and others reflected.

The *direct sounds* travel to the ear with no perceptible delay (unless it's such a huge hall that there is a time lag, for example, between watching a drum being struck and hearing it). In any event, in almost all cases the direct sounds are *the first to reach the ears*. Basically, whenever a sound source is within line-of-sight, direct sounds will arrive before any reflected sounds.

Immediately following the direct sounds are the *early reflections*. These are usually shorter in time lag than the all-important 50 ms mark, and are not heard as distinct echoes. They typically reach the listener after hitting just one or two reflective surfaces. Following the early reflections are *latter reflections*—known as reverberation—which tend to be further and further apart, in terms of time. Unless the room is cavernous, the latter reflections are sounds which have typically hit three or more surfaces.

In a purely random but highly reflective environment, such as a canyon, these tardy arrivals are able to be heard as distinct echoes. If this were the case in our music settings, however, we would have a tough time being able to listen to music without being distracted by echoes. Giant arenas are generally dreaded as concert venues by sound engineers for this very reason.

Fortunately, most concert venues are designed in such a way that there are no discernibly distinct echoes. That is, the acoustics of the hall have been designed to provide continuous reflections to any part of the hall. Some sounds, certainly, may take one second or longer to reach a listener's ears: If these were the only reflected sounds, they would sound as echoes. But the nature of continuous reflections is that they *mask* the impression of distinct echoes. The continuous arrival of sounds to human ears fools the brain into thinking that they are all one sound, when in fact each reflec-

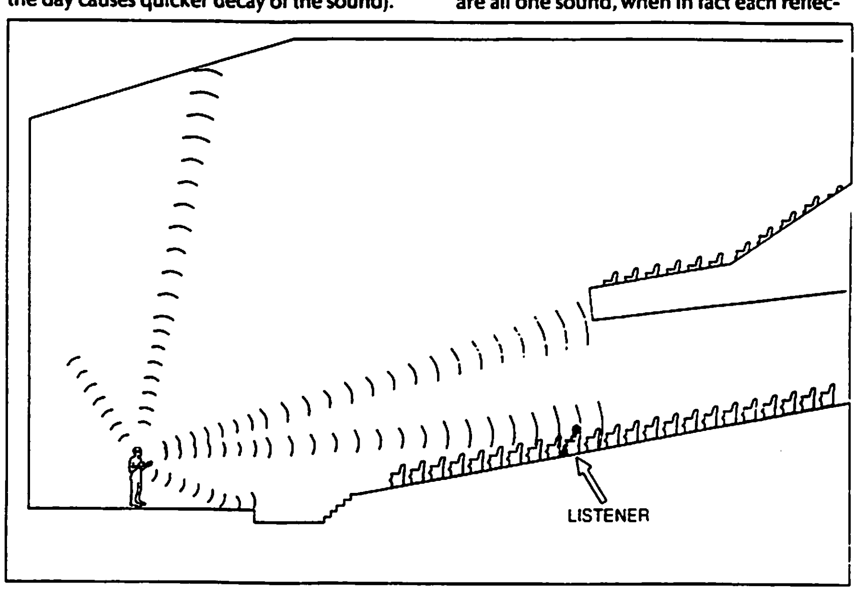


Figure 8.15 In a concert hall, we hear direct—and then reflected— soundwaves.

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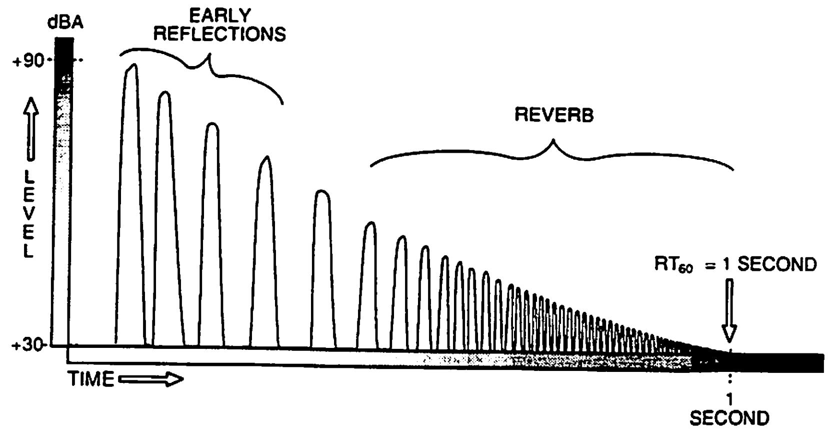


Figure 8.16 Natural reverb can take from many seconds to less than 10 milliseconds to decay. The point at which the reverberant sound decays 60 dB from its first reflection is known as the RT_{60} level. In this case, we have a decay of 1 second.

tion is a component of the entire sound.

If this sojourn into the world of acoustics has been interesting so far, keep reading, since there are a few more advanced concepts to cover. If not, you may wish to skip to page 51. All of these concepts, by the way, are useful in order to operate spatial effects most effectively.

Decay And Acoustic Reverberance. We mentioned earlier the word 'decay.' As we know, all sounds (once they're no longer being produced) eventually decay to an inaudible level. This decay occurs because the sound runs out of what's known as *acoustic energy*. The time by which a reflected sound becomes inaudible is known in acoustics lingo as *RT_{60}*. That term stands for 'reverb time, 60 dB decay,' and comes about from the fact that when a reflected sound has decayed to a level which is 60 dB *below* its original level, it's considered to be inaudible for all practical purposes.

High frequencies reach an RT_{60} level at a *faster* rate than low frequencies—this phenomenon is emphasized when the acoustic space has many highly absorbant surfaces, since those absorbant surfaces affect high frequencies to a greater degree than low frequencies. In addition to reflectivity, long wavelengths travel further than short wavelengths (assuming both were of the initial acoustic energy). Have you ever noticed, when entering a particularly reverberant auditorium or large arena, that the sounds tend to be very 'boomy'—that low frequencies tend to dominate? Well, that's because of this phenomenon of high frequencies having a faster RT_{60} than low frequencies. (This information will all be most useful when we examine reverb units, since there are some reverbs which allow you to have *separate* high and low frequency decay times!)

What are the criteria which determine the reverberance of a space? There are many obscure criteria, all of which make life difficult for acoustic designers, but the main ones are:

- The reflectivity of surfaces.
- The size of the space.
- The angular relationships, or proportions, between surfaces.

Surface reflectivity is in itself affected by two important factors which we mentioned during our 'echo wall' test. The first is smoothness—how flat is the surface? When a sound hits the surface, will it reflect in an even, predictable way, or will it tend to be diffused and scattered by the unevenness? Generally, the smoother the surface, the more reflective, since an uneven surface tends to break up the sound waves. The second factor of reflectivity is the hardness of the reflective surface: Stand right in front of a mirror, at arm's length, and sing or speak directly at it. Then take a thick towel or blanket, and hold it against the front of the mirror. Sing or speak once again. The difference should be dramatic.

The size of the space is the second important criterion. Basically, if we take a large room and a small room, and construct them of the same reflective materials and proportions, the larger room will be more reverberant. This is simple physics: A larger room has greater distances for sounds to travel between reflective surfaces, and hence, longer delay times of the reflections.

The third criterion of reverberance for a space is the angular relationship between its surfaces—its proportions. Our 'two-wall echo' test allowed us to hear what happens with two perfectly parallel walls: Sounds echo back and forth, causing standing waves, as we saw on page 49. A room with standing waves is acoustically very undesirable, since these waves cause an amplification of whichever frequencies happen to be 'standing.' For this reason, concert halls and professional recording studios are never built with any parallel surfaces. Not only are the walls uneven, but so is the ceiling with the floor.

For non-parallel surfaces, the angles at which they meet are also critical for determining a space's reverberance. In fact, the way sound waves interact with the angular relationships of the surfaces is very much like the way a ball behaves on a snooker table. Because there are so many variables for sound, however (including humidity, the number of people in a hall, and even the type of clothes they are wearing), one would generally be better off betting on the outcome of a snooker game than the sonic outcome of a concert hall design!

These are just a few of the reasons why the study of acoustics is as much of an art as a science. Many older churches and concert halls have acoustics which far exceed some of the most modern designs. Not that technology hasn't helped—computer models can help designers predict acoustic responses, and scale models can be built to predict the sound of a hall before it's constructed. Recording studios have also benefited greatly from proper acoustic design: It used to be that the 'deader' the studio—that is, the fewer reflections and reverberance—the better. Now, careful designs can allow the professional studio to fine tune both the control room and the studio to a desired and controllable reverberance.

Basic Psychoacoustics. With the exception of the 50 ms and RT_{60} rules, so far we've been talking about the way sound behaves in space—the study of acoustics—and not so much the way we hear things. Psychoacoustics is the study of the way the brain interprets sounds received by the ears. As recording engineers, it's helpful to understand several concepts of psychoacoustics, in order to use your spatial effects gear to its fullest. These concepts include:

- The *precedent* effect.
- The *relative amplitude* of sounds.
- Our ability to *localize* sounds, due to the effects of reverb, precedence, and amplitude on.

Those of us with hearing in both ears have *binaural* hearing, which allows us to locate sounds in terms of dimension: to the left or right, higher or lower, close or far. In the same way that binocular vision allows three-dimensional visual placement of objects, binaural hearing allows three-dimensional placement of sounds.

Binauaral hearing provides several factors to help us locate sound. The most important factor is that, unless a source is directly in front of us, our ears receive sounds at slightly different times. That is, a sound from the left arrives first at our left ear, then our right. The time difference may seem unbelievably small, but our brains are actually capable of perceiving differences as small as 0.5 ms! (This is not to be confused with perceiving *distinct* sounds, which require a time lag of about 50 ms.) This factor is known as the *precedent*, or *Haas* effect, and it's the most important cue we use in real life to place things in a left to right panorama.

If a soundwave arrives at one ear before the other, it's said to reach our two ears slightly *out of phase*. To go back to sound waves, each wave has a changing *phase*. When we represent a wave on a chart, as we saw in Figure 1.3, the portion on the top of the chart covers the first 180 degrees of the wave, and represents the compression of air; the portion of the wave on the underside of the line covers the latter 180 degrees,

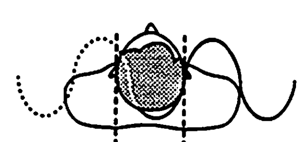


Figure 8.17 A sound wave (in this example, travelling from right to left) generally reaches one ear before the other ear, after which point the brain uses the phase difference as a location cue.

and represents the wave during rarefaction (remember these terms?—if not, see page 2).

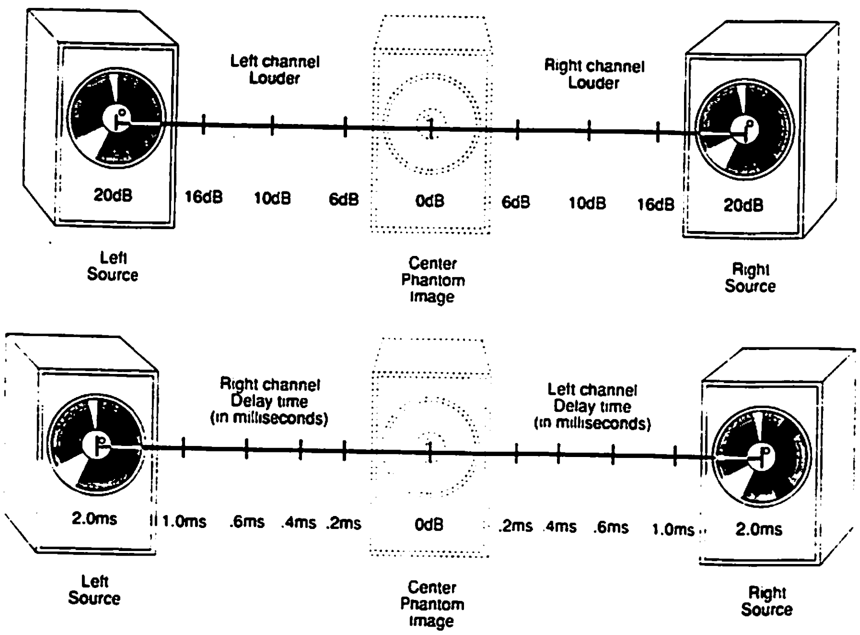
Figure 8.17 shows the wave reaching our two ears at slightly different phase stages. These shifts can be detected by our ears, and are another cue our ears gives our brain to help us locate sounds. As a matter of fact, phase shifters, flangers, and chorus devices are able to simulate a sound out of phase with itself in a constantly changing manner. In this manner, these devices fool our brains into thinking that the instruments or voice is shifting around in space—which is why flanged and chorused instruments sound so 'spacey' and difficult to 'locate.'

Another psychoacoustic factor is *relative amplitude* of sound sources. If we have two speakers, and each speaker is equidistant from our ears as well as playing the same information, whichever speaker is louder will make us think that the sound is coming from that direction. The louder the speaker, the more the image of the sound will shift in its direction. At first, this may seem a lot like the precedent effect, but in fact it's different, since (in the example we've given) each ear is receiving the same information at the same time, in phase.

Our ability to psychoacoustically 'place' objects in space is known as *localizing*, or *imaging*. We localize the *depth* of a sound source, or its relative distance from us, by the degree of reverb associated with it, as well as whatever precedent and amplitude effects may be associated with that sound. Think of it this way: A sound source very close to our left ear will sound much louder in our left ear, and there will be a time lag before the sound reaches the right ear. A distant sound, also on the left, may have a more equal amplitude between the left and right ears, though there will also be a time delay for the sound to reach our right ear. By calculating the amplitude/time delay relationships between near and distant sounds, our brains are able to produce remarkably accurate perceptions of depth.

So far, we've learned some ways that we localize sound sources from left to right, and from front to back. Our ability to localize *vertical* information is much less understood. Some researchers feel that the shape of the outer ear is the key to understanding vertical sound perception; that the angles at which sounds are reflected into our ear canals by the outside ear cause some high frequency phase shifting, by introducing slight delays between the sounds that reach our ear canals. It's felt that these delays place sounds within the 'up and down.'

With regular stereo loudspeaker reproduction, it's very difficult to create vertical imaging. The popular chorusing effect that's used a lot with guitars and keyboards, however, seems to create some sense of vertical sound movement. This may have something to do with the fact that they cause some higher frequency phase shifting—thus suggesting that time delays *do* play a part in vertical localizing.



Figures 8.18a, 8.18b When two equidistant speakers play the same information at the same level, we perceive a 'phantom' sound image that's in the center. Should one speaker's sound become quieter or be delayed, we'll perceive an image that shifts towards the other speaker. The delay effect—known as the 'Haas effect'—is something to bear in mind when using a digital delay.

When we get our hands on some nifty, inexpensive digital reverb, and go about programming it to simulate acoustic spaces, an understanding of acoustics and psychoacoustics will be indispensable. Many of the latest reverbs, even in the $500 range, allow us to control separately the high and low frequency decay rates (RT_{60}s), as well as the level of early reflections, the diffusion of the surfaces, and much more. A quality reverb is very useful for creating both natural and unnatural acoustic spaces.

Similarly, the digital delay line (DDL) has become a very important studio tool, not only for echo effects, but chorusing, flanging, and more. Understanding how echoes behave, and at what point they become indiscernible as echoes, is critical information for maximizing your potential with a DDL. So, without further ado, let's move on to some of these wonderful devices.

Delays

The first artificial delay was created by that early master of the magnetic tape recorder, Les Paul. He found that a single echo could be heard by monitoring the playback head of a 3-head recorder, while also listening to the source being recorded. If the output of that playback head was fed back into the record head, multiple echoes could be heard and recorded.

For many years this and other types of *tape echo* was the only way to create echoes and other delay effects artificially. By varying the speed of the tape, the timing of the echo could be somewhat regulated, and by varying the amount of *feedback* from the playback head back to the record head, the number of repeat echoes could be controlled. Adding more playback heads would yield delays with different times. From the end of the 1960s and into the '70s, many studios had self-contained tape echo units, such as the Maestro Echoplex and the Roland Space Echo.

By the mid-'70s *analog* delays became popular as the first tapeless delays; a number of these are still available as foot pedal devices. In an analog delay, a circuit known as a *bucket brigade* replaces tape. This circuit can electrically store audio signals for adjustable periods of time before playing them back. They're called bucket brigade circuits because they pass the signal along, like a fire brigade passes a bucket of water, until the signal is *tapped* and played back at some point along the line. While more convenient to use than tape delays, and ultimately cheaper, analog delays have a relatively poor signal-to-noise ratio, and poor high-end frequency response. Because of this, many professional engineers continued to use tape as the delay of choice, until the advent of *digital* delays.

First available in 1972, the digital delay (or *DDL*, for *digital delay line*) broke the $1000 barrier and became affordable to smaller studios by the early 1980s. DDLs are usually available for a heck of a lot less money than any of the tape delays, and offer much greater versatility and performance. Most DDLs can perform:

- Single and multiple echoes.
- *Doubling*, which simulates a 'doubled' sound by means of a very short single echo. This can also serve as a simulated stereo effect.
- *Chorusing*, *flanging*, and other related sounds. There are separate chorus and flanging devices available as foot pedal effects, but most self-contained DDLs offer superior performance and control for these effects.

Some of the more advanced DDLs have additional features, such as:

- Stereo delay effects, by means of two separate delays feeding two separate outputs of the DDL.
- Programmability, which allows the storage and recall of program settings into the unit's memory.
- MIDI control, which allows program changes and other adjustments to be

made from external MIDI sequencers and instrument controllers (see page 106 for more about this).

- *Sampling* functions, where a digitally-stored audio signal (such as a drum hit or—with some sampling DDLs—a signal as long as a song's chorus) can be held indefinitely and then replayed, or 'triggered,' with a switch or footswitch.

Let's take a look at some of the controls found on various DDLs and examine how they can be used to create some of these effects.

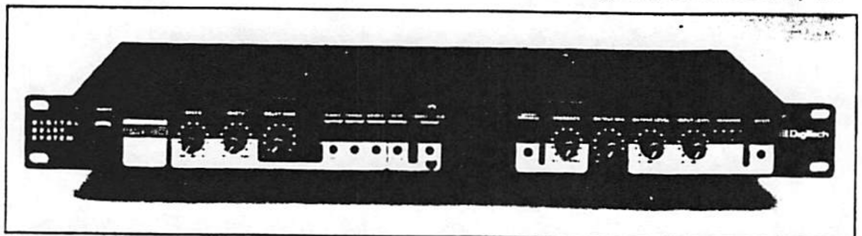


Figure 8.19 The Digitech RDS1900 digital delay offers up to 2 seconds of delay, and can perform chorusing and flanging effects.

DDL Controls

Digital delays use analog-to-digital converters (ADCs) and digital-to-analog converters (DACs), which we will learn more about on page 125. Simply, a DDL uses an ADC to convert incoming audio signals to numbers. These numbers are then held electrically in memory for the desired delay time, after which the DAC reconverts the numbers to sound, which is spit out as a delayed audio signal.

As with any digital processor, it's important to not overload the ADC. The result is usually much worse than the type of overload distortion heard with analog devices, and is best described as digital 'glitching.' Consequently, most DDLs have an *input gain* control and meter. Remember to use the meter to keep the input signal as high as possible (to maximize the signal-to-noise ratio) without glitching or distortion.

Other common input/output (*I/O*) controls include:

- An *effect bypass* switch. This is useful for comparing the original, or 'dry' signal, with the delayed, or 'wet' signal.
- A *wet/dry* mix control. If an instrument is plugged directly into the effect, this control should be set to your sonic taste, or to 50% for flanging or chorusing. If the unit is connected in an effect send/return loop with a mixer, the mix control is normally set to 100%, so that the only output of the effect is the delayed signal; the balance of the wet and dry signal is then determined at the mixer.
- An *inverse phase* switch puts the wet signal 180 degrees out of phase with the dry signal. Some DDLs have a separate out-of-phase output jack—but beware: When two out-of-phase delayed output signals are summed to mono, they'll cancel each other out. The moral: If your music may wind up in mono (AM radio, TV/video, etc.), don't record in- and out-of-phase signals together!
- Some units have *level matching* switches, to accomodate different level gear (see page 19).

The *delay time* control is often accompanied by a digital readout of the delay setting in milliseconds. The control itself allows you to adjust how much time will elapse between the original sound and any subsequent delayed signals, and is either a ten-key pad (like a telephone, to allow direct programming of the delay), an incremental-adjust (usually two keys, + or –), or a switch/knob arrangement. At minimum, most DDLs offer about 200 ms of delay, though better units can have ten or more seconds of delay memory.

Remembering that the shortest delay we can perceive is about 50 ms, echoes can be created at this setting and up. Below 50 ms, a DDL is used to create chorusing, flanging, and reverberant early reflections.

If you know the tempo of a song, it's possible to calculate 'synchrosonic' delays which fall on the beat. Simply divide 60,000 (the number of milliseconds in one minute) by the tempo (in beats per minute), and you'll have a setting in milliseconds which will yield delays that match the beat.

A *feedback* control adjusts how much of the delayed signal is fed back into the digital delay, in order to create repeating delays. When the feedback control is set at 0, and the delay time is greater than 50 ms, you'll hear just one echo. As you increase the feedback level, you'll hear an increasing number of echoes, which gradually fade out. Through careful adjustment, you can set anywhere from two to 30 or more gradually decaying echoes. A *hold* switch or footswitch will cause an echo to repeat *ad infinitum*, until disengaged.

When the delay time is less than 50 ms, the feedback control is used to adjust the 'intensity' of the chorusing, flanging, or early reflections. The more feedback, the more noticeable the degree of the effect. But be careful with this control: Too much feedback can create a runaway situation, causing a *feedback loop* (similar to microphone feedback), which is both horrible to hear and potentially nasty to your speakers and other equipment.

Most DDLs are able to *modulate* the delayed signal. Modulation is usually heard as a slow, steadily changing sound, and is necessary to create chorusing and flanging effects. To get modulation, DDLs use an adjustable LFO (low frequency oscillator), which varies the delay time by slight but noticeable degrees. The speed with which the modulation takes place is controlled by a *speed* or *rate* control; the depth of the modulation is controlled by a *depth* or *width* control.

Normally, modulation speed is set quite slow. If the speed control is set too high, the effect can become ridiculous—great for making instruments and vocals sound as if they're 'underwater.' The depth control should be set low if just a subtle flanging or chorusing is desired, and higher for a more intense effect.

Modulation is typically used at delay times of less than 50 ms. If it's used at higher delay time settings, such as 500 ms or more, the effect can be fairly bizarre, with strange pitch bends and the like.

Flanging And Chorusing

Flanging and chorusing are true spatial effects—they cause guitars, keyboards, and other signals seemingly to shift in space and swirl around. Wearing a pair of headphones and listening to a heavily flanged or chorused stereo sound can actually disturb your equilibrium! Whether you use these effects to create a shimmering guitar or a full-bodied bass, the fundamental concept behind them is the delay of time. The actual delays are too fast for us to perceive as distinct echoes: Flanging is in the 0.2 to 20 ms range, and chorusing uses delays of about 15 to 35 ms. The modulation control is the key to creating the effects, however: It introduces a slight shifting of these times, and as our brain tries to follow the shifting it gets confused, causing us to think the sound is moving.

Many guitarists, bassists, and keyboardists use subtle chorusing almost all the time, to 'thicken' their sound and give it some motion. Flanging tends to be more of a special effect, since its intensity can become a bit wearisome, and the effect can make instruments sound a bit thin. Flanging gets its name from an old trick involving tape: If an audio signal is split, and recorded onto two tape decks at the same time, listening to the playback heads of those two decks should in theory let us hear two identical signals. In practice, however, two un-synchronized tape decks will run at slightly different speeds. This speed variation will cause the sound to shift when listening to the two decks, as small and slightly changing delays occur. A popular technique to induce this effect to a further degree was to apply thumb pressure to one of the tape's flanges (metal reels). By 'flanging' in this manner, the time delays would increase—and by varying the pressure, the sweeping, shifting sound would change over time. In the modern DDL, the modulation control replaces the thumb, and the delay time replaces the tape speed variations.

Phase shifting, or phasing, has dropped in popularity since the mid-70s. We've spoken about how changes in a soundwave's phase is known as phase shifting. This is often an undesired sound, caused by misalignment of tape heads, or by poor equipment design. The induced effect of phase shifting sounds very similar to flanging, although it's a bit 'fuller' sounding. Instead of manipulating the timing of delayed signals (as with flanging), phasing manipulates the actual phase of the audio signals. None of the DDLs made are designed to do true phase shifting, but some of the foot pedal phasers, most commonly those by MXR, can still be found.

On some of the more advanced DDLs, the modulation *waveform* is selectable. Normally, the modulation follows a smooth-sweeping sine wave pattern; some DDLs allow you to choose a sawtooth or even square waveform, creating rapid-changing modulation effects, that tend to have a 'full-on,' 'full-off' type of sound. Some better units also have a *clock* input, which allows the speed of the modulation LFO to be controlled independently by a drum machine or sequencer.

Reverbs

Reverberation, as we've seen, is a complex collection of acoustic reflections, which are indiscernible as distinct echoes. Reverb is a common sound in nature and in architectural structures—but until recently it's been very difficult for the home recordist to get a good reverb sound.

People who record classical music in a good concert hall enjoy the benefit of the natural reverberation in the hall. Typically, live classical recording takes place with two microphones some distance from the instruments, so that the sound of the hall is recorded along with the direct sounds of the instruments.

Many commercial recording companies invest in quality studio acoustics, and also record the sound of the studio along with live instruments. Most home and quite a few professional recordings, however, are either made directly—with synthesizers, drum machines, and other sound sources plugged directly into the console—or made with close-miking techniques—where the mike is placed very close to a singer or instrument. Because of direct and close-miked recording, and with many of us forced to be content with the acoustics of a bedroom or garage, there's not a lot of natural room reverberation being recorded these days.

Without natural reverberation, we need to depend upon *artificial* reverberation to give our recordings depth, and the sense of acoustic space. Without any reverb, recordings sound very up-close and 'dry'; while this can be effective with some songs or music styles, many people find reverb-less recordings lifeless, and two-dimensional.

There are four types of artificial reverb devices:

- Acoustic reverb chambers.
- Plate reverbs.
- Spring reverbs.
- Digital reverbs.

Let's take a look at each of these types . . .

Acoustic Reverb Chambers. The idea behind an acoustic reverb chamber is simple: Take a highly reflective room—anything from an elevator shaft, to a long hallway, to a squash court could do the trick—and place inside of it a loudspeaker and a couple of microphones. Feed the speaker with a signal from the mixer (any track or group of tracks to which you want to add reverb) and let that signal reflect around the room. Then pick up that reflected signal with the microphones. Finally, mix this reverberant signal with the original dry signal, and *voila*—artificial reverb.

Reverb chambers have been around since the early days of recording, and a fair number of professional studios still boast some sort of chamber. Assuming one has the gear and the room, there's no reason why a home studio couldn't have its own chamber. Fortunately, for the apartment dwellers and others among us, there are several other, more versatile, ways of getting artificial reverb, all of which are self-contained.

Plate Reverbs. Even in this day of digital-mania, some of the most coveted reverb units are plate reverbs.

Like the chamber, a plate works on the idea of broadcasting a dry signal, delaying it, and picking it up again. But instead of a room, a large, thin, metal plate is used; instead of a speaker, a transducer (like a small speaker) is used to broadcast. The plate's dimensions can be anywhere from 0.5 meter x 1m (1.5' x 3') up to 3m x 4m (9' x 12'). The broadcast transducer is located at one end of the plate. The transducer receives the signal to be processed from the mixer, and as it vibrates, it causes the plate to vibrate. When this happens, all sorts of random vibrations travel throughout the plate, until they are eventually picked up by a couple of contact transducers, and fed back to the mixing console.

Most plates have a mechanical means of 'damping' the vibrations of the plate, which allows you to adjust the decay time. Typical decays are quite short, less than a second, though some of the larger plates can provide several seconds of reverb.

The 'plate sound' is a classic reverb sound, popular for drums: very 'tight,' and bright sounding, with a quick decay. The frequency response of most plates is wonderful, covering the complete audio spectrum. In fact, many professionals look for a great simulated plate sound when they audition the latest digital reverbs.

The disadvantages of plates include their size, as well as their susceptibility to picking up external noises (the plate is very sensitive, and can act like a giant microphone). Because of the latter, plates must be isolated from external sounds, including loudspeakers. This means either mixing with headphones, or placing the plate in another room.

Plates, moreover, have the same limitations as reverb chambers and spring reverbs: They sound the way they sound, and not much variation is available. Plates can and should be tuned, but even the least expensive digital reverb has more variations in sound. Nevertheless, if you have the space, there are some good buys out there in the way of used plates, as the digital revolution plows onward. In fact, building your own plate used to be a popular engineer's pastime—it's not all that hard, and it can be quite inexpensive.

Plates are still commercially available, from Echoplate/Studio Technologies, and other manufacturers. The finest of these plates, such as those made in Germany by EMT, have plates made of thin gold foil, can set you back several thousand dollars, but sound great.

Spring Reverbs. Until recently, spring reverbs were the mainstay of affordable reverb devices. Still found in guitar amplifiers, spring reverbs can be quite fine for many applications. The technical concept remains roughly the same as rooms and plates: dry sound amplified and delayed, then picked up. Instead of a plate, however, two or more springs are used to induce the delay (a single spring would tend to resonate on a particular frequency).

At one end of the springs, a transducer applies the dry signal, which travels up and down, bouncing around the springs. The wet signal is picked up by one or two transducers at the opposite ends. Unlike rooms and many plates, spring reverbs usually have some type of dry/wet mix control, so that the output of the reverb can be adjusted between an unprocessed/processed mix. Most springs are mono, though some have stereo outputs, which improve the realism of the reverb signal.

Some of the best spring reverbs are made by AKG and employ a mechanical damping system for the springs, so that the decay time can be adjusted. Many springs have built-in EQ sections, so that the tone of the reverberated signal can be adjusted. At their best, even inexpensive spring reverbs can sound very good with vocal and other non-percussive sounds. They offer smooth and even decay, and excellent frequency response. At their worst—with drums and other sharply percussive instruments—springs can sound horrid, unless used very carefully. The reason is that when the spring receives a signal which has too much 'attack,' the spring is overdriven, and produces a metallic 'boing' sound. With drums and other such instruments, the levels must be carefully adjusted, and the dry signal should be EQ'd to reduce the punchiest frequencies.

Spring reverbs are also susceptible to outside noise and vibrations, though to a much lesser degree than plate reverbs.

The future of the spring reverb is not especially bright. There are already digital reverbs which far surpass the performance of springs, offer many more sounds, and cost less. All the same, if you already own a spring, or are able to get one cheap as a secondary or tertiary unit, great. Used sparingly and carefully, springs can provide satisfying performance, particularly with vocals, woodwinds, and guitars.

Digital Reverbs. Imagine this: It's 1978, and you're an engineer at a top L.A. studio. Everything is state-of-the-art, from the automated mixing console to the 24-track, right on down to the gold-foil EMT plate reverb, the acoustic reverb chamber, and the AKG spring reverb. Then one day a large, heavy box arrives, brought by a factory representative from Lexicon, a company in Massachusetts. Inside, you're told, is a new reverb unit, the Lexicon 224—a *digital* reverb unit. . . .

. . . The rep explains how there are no moving parts, no plates, no springs, nothing. Just lots of integrated circuits, like the Lexicon PCM 41 DDL in your rack. He tells you how the incoming analog audio signal is converted to a series of digital codes, which can then be manipulated in a bunch of different ways, and then reconverted to an analog signal for playback. All very nice, but what do reverberant manipulated numbers sound like? The rep plugs it in, you connect it up to the mixing console, put on a multitrack tape, and turn up the control room listening level. Remote control in hand, the rep selects a 'Plate' setting, and you listen to it . . .

. . . Sounds nice, almost as good as the EMT, but the EMT costs half the price. While you keep this to yourself, the rep selects 'Hall.' All of a sudden, your drum tracks sound as if they're in a giant concert hall, a much bigger sound than you were ever able to get from the acoustic chamber, let alone the plate or spring. You watch as the rep asks you how big, in cubic meters, you'd like the hall to be! As he moves the controls from a large hall to a small chamber, the reverb begins to collapse into itself, as the simulated walls get closer, creating an intimate yet very alive sound. Before you can express your amazement, he asks you what sort of high frequency decay you'd like, and how many early reflections. But his questions fly past you, as you start to concentrate on ways to sell this *wunderbox* to your boss. *Hmm, maybe if we sell both the plate and the spring, the studio may be able to cough up the 12 grand this thing costs . . .*

It's still possible to spend upwards of $10,000 on a digital reverb, but what's more amazing is that for well under $500, you can get one which in many ways *outperforms* the first digital reverbs. Clearly, digital reverbs are here to stay, and they are undoubtedly the reverb of choice for the personal recording studio.

The digital domain offers the ideal way of manipulating audio signals. Armed with a thorough understanding of psychoacoustics, the digital reverb designers create *algorithms*—mathematical formulae, which are used for manipulating the digital information to simulate both real and surreal acoustic spaces.

As with digital delays, digital reverbs can create numerous different effects, all available with the touch of a switch. In their simplest incarnation, they offer the choice of several different settings, from simulated plates to large halls, as well as a few special effects which we'll examine shortly. The Alesis MicroVerb II is a good example of this type of reverb, with 16 different presets accessible via a front-panel knob, and stereo outputs.

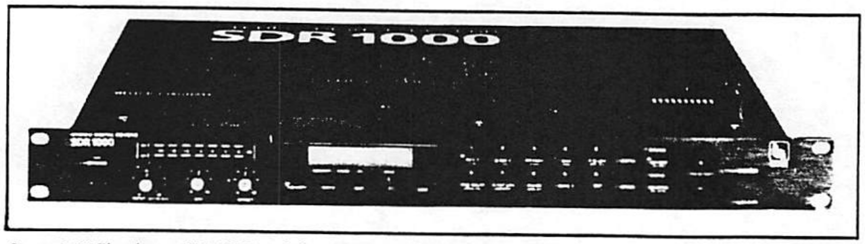


Figure 8.20 The Ibanez SDR1000 is a fully programmable digital reverb. It can function as a stereo reverb or as a dual-channel device, with each channel capable of different reverb or delay effects.

Some preset reverbs allow their settings to be called up remotely via MIDI. The ART ProVerb is one such reverb. It has 99 preset programs, all of which can be called up either via a front panel switch or MIDI (we'll examine MIDI effects on page 109).

What if you want to create your own reverb effects, rather than using factory presets? The most recent incarnation of the original Lexicon 224, the Lexicon 480L, costs over $8000, but there are fully-programmable digital reverbs for well under $1000. These units don't do everything that the 480L does, still they do allow you to choose between a number of different settings, including plates, halls, small rooms, and several special effects. In addition, various parameters of these sounds can be altered such as decay times (for both high and low frequencies), the number and level of the simulated early reflections, as well as pre-delay and other settings.

Digital Reverb Controls

Unlike DDLs, very few digital reverbs share the same parameter controls. The formulae used to create artificial digital reverb are much more complex than those used for delays. Because of this, manufacturers choose different ways of changing similar functions.

Some reverbs are dual-channel devices. These units allow you to select completely different parameters for each of their two channels. For example, with the Ibanez SDR 1000+, you can have a simulated plate sound on one channel, and a large hall on the other. This Ibanez unit, in fact, can perform delay functions on one channel while performing reverb on the other. This dual-channel function shouldn't be confused with stereo operation—many digital reverbs have stereo inputs and outputs, but only one processing mode can be selected at a time, and that same effect is applied equally to the left and right stereo channels.

Here's a listing of commonly-found digital reverb parameter controls.

Input And Output Level Control, Input Level Meter, And Wet/Dry Mix. Just as on the DDL, these allow the user to adjust the signal levels coming into and leaving the reverb. Some units, such as the Lexicon 200, allow for separate Left/Right input and output level adjustments.

Pre-Delay. As we learned back on page 49 (Figure 8.15), in most real-life settings—such as a concert hall—the direct (dry and unreflected) signal is the first to reach a listener's ears. Following this direct signal, after some delay, are the early reflections, then the reverberant signals. A pre-delay control is a means of controlling the length of time which passes between the dry signal and the first reflections. If offered, the range is typically from a few milliseconds up to a half second or longer. Most of the preset digital reverbs (those without adjustable parameters) have preset pre-delays built into their various programs, in order to enhance the realism of the programs.

Aside from adding to the realism of reverb programs, pre-delay can be used as a special effect. For example, the timing of the pre-delay can be set to match the beat of the song, so that the delayed reflections happen a quarter- or an eighth-note after the dry signal. The formula which correlates delay times with song tempo is given on page 52.

Room Type. The room type describes how many reflections take place, how complex those reflections are, and other similar functions. This allows you to select the basic reverb sound: The simulated 'room' in fact, can sound like anything from a small casket to a plate reverb, from a mid-sized room to the Grand Canyon. When designers of digital reverbs sit down at the drawing board (or design computer), it's this area which will take up most of their mathematical musings—as they create the room algorithms which ultimately lead to the reverb's distinctive sounds. Most digital reverbs offer anywhere from two to 30 or more different room types. Dual-channel units will allow you to choose one or two of these rooms at a time, while most stereo units limit you to one room type at once.

The room type is an important choice when recording. Part of the fun, however, is that there are no rules—much of modern music benefits from a lush mixing of different sonic and reverberant textures. So with a bit of experimentation, 'plate' drums, along with a 'small room' guitar and a 'chamber hall' synthesizer could be one of many satisfying room combinations.

Room Size. This control allows the physical volume of a simulated room to be changed. It's generally expressed either in cubic meters, or in terms of the distance in meters from the 'stage' (dry sound source) to the far wall. Whereas the room type control changes the types and angles of reflections, room size alters the time it takes for sounds to travel throughout the room, before and after they reflect. A well-designed reverb can truly make a room change in size, so that we are psychoacoustically tricked into hearing larger and smaller spaces.

Early Reflections. On some reverbs, there are two 'ER' controls: One for the *density* of the early reflections, and the other for their amplitude *level*. Again, early reflections are the first echoes our brain picks up, before the 'wash' of densely-packed reverberance. By changing the density and level of the ERs, their sound can change from distinctly audible echoes to very close, quickly reflecting sounds. In real life, it's difficult for us to perceive these early reflections from the ensuing reverb, so these are difficult sounds to imagine without having heard them.

Diffusion. Not found on all programmable reverbs, diffusion allows you to control how *diffuse* the reverberant reflections are. High diffusion levels will create very mixed and indistinct reverb sounds; low diffusion will allow the reflections to become more audible as distinct echoes.

Decay Times. In all programmable digital reverbs, there is at least one decay control, which allows you to adjust how much time passes from the first reflections until the processed signal decays to the RT_{60} level (page 50). Decay times range anywhere from about 10ms to 99 seconds, depending upon the unit. If you want a 'natural' decay, typical real-life rooms have from 10ms to three or four seconds of decay.

Many programmable reverbs allow you to establish *separate* low and high frequency decay times. When there is a means of adjusting the high frequency decay,

there's often a *crossover* frequency adjustment: This allows you to choose at which frequency the low band ends and the high band begins (the choice is usually from about 2 to 8 kHz).

In most real-life settings, it's rare to hear reverberant frequencies much higher than 10 kHz, and frequencies above 6 kHz don't tend to last much longer than half a second or so. For this reason, when simulating a real setting, the high frequencies should decay appropriately (an equalizer can be used if there isn't a high-frequency decay control). There's no law, however, that says you can't create surreal, synthesized acoustic spaces —your decays can have any setting you want. When just low frequencies are set at a long decay, the space can sound long and narrow, like an oil pipe or sewer. This is particularly effective with kick drums and lower frequency percussive sounds. On the other hand, if only the high frequencies are given a long decay, the sound becomes ethereal, and makes instruments seem as if they're floating or suspended in air—this is great with electric guitar, and for certain vocal special effects. Do keep in mind that long decay settings can make the sound a bit jumbled—for this reason you may wish to have your decay decrease as the song's complexity increases.

Reverb EQ. With many reverbs you'll find some type of built-in EQ, either programmable or not. A built-in EQ is useful, since it allows you to tailor the EQ of the reverberated signal without tying up any outboard or mixer EQs. The options are vast, and—as with all of these parameter controls —use your own taste to find the right setting. For reasons related to natural high-frequency decay, cutting the higher frequencies will tend to make the reverberant setting sound darker, less reflective; boosting higher frequencies and cutting lower and mid-frequencies will help create a bright, very alive space.

Some reverbs, rather than offering a true EQ, have high-and lowpass filters. Still others offer a control known as high frequency *damping*—a cross between a lowpass filter and a high-frequency reflectivity control. Damping softens the high end while often simulating fewer reflections.

Special Effects. As part of the different room selections, most digital reverbs offer at least a few special effects settings. The three most common special effects include *gated*, *inverse*, and *infinite* reverb.

The gated reverb sound was first made popular by British producer Hugh Padgham, with artists Phil Collins and Peter Gabriel. The sound was originally created by taking a long decayed reverb sound of a drum, and *gating*, or quickly cutting off, the reverberant sound before it's had a chance to decay. It's a very distinctive sound, rather like the sound of a huge, exploding drum which disappears instantly. While Padgham and many others have created this sound using a separate reverb and gate (which we'll examine on page 57), many reverbs have this sound programmed internally, often with an adjustable gate time. While gated reverb is a great sound for drums and bass guitars, it can become a bit wearisome if overused.

The inverse sound is pretty wild: It's reverb in reverse! Basically, the sound starts with a quiet decayed signal, and builds up in volume and intensity until it suddenly stops. The effect isn't too unlike a train coming closer and closer, until it reaches you and suddenly disappears. Great for sound effects (alien landings and so forth), but of limited musical use.

Infinite reverb is a classic space age/new age music effect. Some reverb manufacturers have referred to it as the 'bottomless' room. Sounds which enter this mode reverberate indefinitely. With careful use of the front panel switch or footswitch (to 'open' and 'close' the door of the infinite room), you can build entire choirs by singing one note at a time; previously sung notes will just keep going forever, until you switch the effect off. Great for just about any instrument or voice, though the sound can become garbled and muddy if too many notes, or any repeated percussions, are played at once. The notes which enter this room should be harmonically complementary, unless you want a dissonant sound (*i.e.*: if you're in the key of C, notes C, E, and G will create major melodic results, C, E♭, G, and B♭ will create minor-7th results, and so on).

Compressors And Other Dynamic Processors

Compressors, limiters, and gates are the best-known of a group of signal processors known as *dynamic processors*. Dynamic processors do what their name suggests: They control the amplitude levels—the dynamic range—of audio signals. Depending upon which device, and how it's set, a dynamic processor can:

- Prevent signals from overloading inputs or tape, or from destroying loudspeakers.
- Reduce or eliminate noise from an instrument or mike.
- Increase the apparent sustain of instruments which decay, such as bass guitars.
- Remove unwanted sibilance ('sss' and 'shhh' sounds) from vocal tracks.
- Increase the apparent volume of a complete signal.
- Be used with reverb to create popular special effects.

These processors can be placed into two broad camps, those which *reduce* dynamic range, including:

- *compressors, limiters, de-essers,* and *duckers*

and those which *increase* dynamic range, including:

- *expanders* and *gates.*

Some of the planet's better mixers include these units as built-ins to each channel, much the same way as they would include EQ; this will certainly be a growing trend, as mixers become more powerful and less expensive. In the meantime, most of us will be using outboard, rack-mountable dynamic processors. As we'll learn, some of these devices can perform many different dynamic functions, while others are built with one application in mind. (Some manufacturers offer foot pedal dynamic processors, such as compressors and noise gates; while this chapter will also apply to them, they're generally limited in the way of functions and controls.)

Let's begin with a look a compressors and limiters, which will serve as good models for understanding all of these dynamic processors.

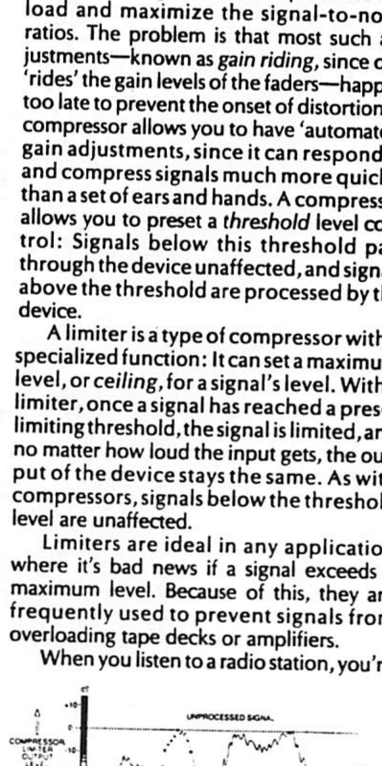
Compressors And Limiters. Compressors reduce the overall dynamic range of the signal, by making loud passages quieter; that is, they compress. Since the total range of level between the loudest and quietest passages is reduced, a compressor's net effect is to make quiet passages appear louder—at least in relation to the loud passages. Compressors can be used to make the volume of a dynamically-uneven signal more consistent—such as that from a vocalist who may have trouble singing at consistent levels.

As an engineer, there are lots of signals you'll encounter, such as vocals, which have widely varying levels. When recording or mixing down these kind of signals, you'll have to adjust fader levels in anticipation of the signal changes, in order to prevent overload and maximize the signal-to-noise ratios. The problem is that most such adjustments—known as *gain riding*, since one 'rides' the gain levels of the faders—happen too late to prevent the onset of distortion. A compressor allows you to have 'automated' gain adjustments, since it can respond to and compress signals much more quickly than a set of ears and hands. A compressor allows you to preset a *threshold* level control: Signals below this threshold pass through the device unaffected, and signals above the threshold are processed by the device.

A limiter is a type of compressor with a specialized function: It can set a maximum level, or *ceiling*, for a signal's level. With a limiter, once a signal has reached a preset limiting threshold, the signal is limited, and no matter how loud the input gets, the output of the device stays the same. As with compressors, signals below the threshold level are unaffected.

Limiters are ideal in any application where it's bad news if a signal exceeds a maximum level. Because of this, they are frequently used to prevent signals from overloading tape decks or amplifiers.

When you listen to a radio station, you're



Figures 8.21a, 8.21b, 8.21c Compressors reduce the level of signals which pass a threshold, while 'hard' limiters prevent those signals from exceeding the threshold.

listening to compressors and limiters in action. Radio stations use compressors in order to sound 'louder': By making all the quiet passages seem louder, the apparent overall volume of the station is increased, so that it stands out on the radio band. Depending on how commercially-minded the station is, different amounts of compression are used; if you ever hear an announcer sounding particularly 'breathy,' or 'squashed,' with a volume level that sounds the same if he or she is whispering or shouting, you're hearing a lot of compression. Similarly, radio station use limiters to prevent *overmodulation*—operating at too high a broadcast level. Overmodulation is federally prohibited, since it can cause interference on an adjacent wavelength.

When you listen to most analog records, you're also hearing compressors and limiters at work. As we know from Chapter One, the dynamic range of human hearing is about 150 dB. Most live music covers a dynamic range of about 80 to 100 dB, from the quietest to the loudest passages. Classical music, in fact, generally has the widest range; rock music usually has a narrower range. But even 80 dB is more than an LP record can accommodate: Using the quietest vinyl, a record has a dynamic range of 60 to 70 dB—from the quietest audible passage to the loudest peaks before distortion. In order to 'squash' 80 to 100 dB worth of dynamic range into 60 or 70 dB, record mastering labs use compressors. In order to prevent any accidental, distorting peaks, they use limiters.

While there are separate devices available, for the most part compressors and limiters are found as one combined unit, called a *compressor/limiter*, or *comp/limiter* for short. Such units, made by Furman, Symetrix, dbx, Fostex, Orban, Yamaha, and many others, are designed to perform either compression or limiting, and a few will perform both functions simultaneously in the same channel. Many of these are 2-channel devices, allowing you to perform compression in one channel and limiting in the other. Two-channel comp/limiters can operate in stereo, as well: By selecting the second channel to 'slave' from the first, the controls of channel 1 also control channel 2, and the two channels respond identically.

Most comp/limiters use *VCAs*—voltage controlled amplifiers—to adjust gain. VCAs respond to varying voltages, such as those from an input signal. With comp/limiters, when the input signal exceeds threshold the VCA begins to decrease gain. The *amount* of gain reduction is set by a *ratio* control.

Why a 'ratio' control, rather than an 'amount' control? The reason is because dynamic processors measure the amount of their effect as a ratio of the unaffected input level to the processed output level. Let's consider a compressor. If we set its ratio control to a 1:1 setting, for every decibel that the input level increases or decreases, the output level also changes by 1 dB. Consequently, if we sang through a mike into a compressor with a 1:1 ratio, we'd hear no change when comparing the input to the output of the device.

If we set the ratio control at 2:1, however, it takes 2 dB of input level change to cause a 1 dB output level change. If for example, our singing level increased by 6 dB, we'd only hear a 3 dB change. In other words, our mike level would be compressed at a 2:1 ratio.

Remember, compression works for both increases and decreases in level, so that if we sang 6 dB quieter, we'd hear the mike level decrease by only 3 dB (assuming the input was still above threshold). As you can imagine, a 2:1 compression ratio can help even out the mike levels of an inconsistent singer—all level changes would be reduced by 50%.

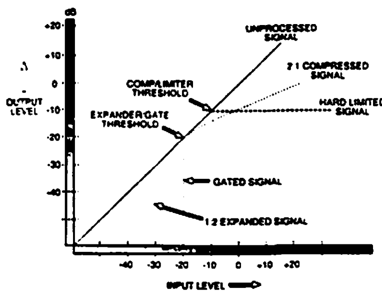


Figure 8.22 Signals which exceed a preset threshold of a compressor or limiter are processed. With expanders and gates, however, signals which drop below a threshold are processed, as indicated. Below that threshold, expanders will make quiet signals even quieter, and gates will virtually silence them.

Compressors usually operate with ratios of 1.5:1 to 10:1. At levels above 8:1 or so, the compressed signal can sound very 'squashed,' since it takes an input level change of 24 dB or greater to result in an audible output change of 3 dB. At those high settings, all signals above the threshold level tend to sound close to the same volume, whether they were originally quiet or loud. (Depending upon the desired effect, compressors can be set with high or low thresholds, as we'll see shortly.)

Limiters operate at compression ratios of 10:1 or greater. If your compressor offers a 10:1 or higher ratio, it can behave as a limiter, and is in fact a comp/limiter. Here's how this ratio setting turns a compressor into a limiter: At high ratios, there's so much compression that no matter how high the input signal becomes (once it has exceeded the threshold), there's no audible level increase at the output. For example, at a ratio of 20:1, it takes an increase of 60 dB to raise the output level by just 3 dB.

So, for example, let's say we wanted to limit a miked vocal. The first step is to have the vocalist sing alternately loud and very loud passages. We could then adjust the threshold control so that only the very loud passage would be affected by the limiter. Consequently, at levels below threshold the vocals would not be limited. Above the threshold, no matter how much louder the vocalist sang, there would be no discernible change—a very useful way to prevent input overloading.

Some limiters allow ratio settings of 40:1 or even ∞ :1. When you have settings this high, the device can function in what's known as an audio 'brick wall,' and output level changes are completely stopped.

In Figure 8.22 notice the points on the chart at which the processing engages. With the compressor, that point is slightly rounded, whereas with the others the point is sharp. These processing points represent, respectively, *soft-knee* and *hard-knee* processing. Many compressors are soft-knee, where the introduction of processing is somewhat gradual and less audibly noticeable; dbx calls this 'Over Easy' compression. Given the nature of limiting—avoiding instantaneous overload and distortion—most limiters engage instantly, creating a 'hard-knee' on the graph.

Aside from ratio and threshold controls, most comp/limiters have *attack* and *release* controls: These are used respectively to adjust how quickly the compression or limiting takes place, and how soon it stops. We've already said that signals which exceed the threshold are processed. The attack control adjusts how much time passes once a signal has exceeded the threshold before the signal is processed (signals which are below the threshold setting are not processed). The release control determines the time it takes for the processing to stop once a signal has dropped below the threshold level. Remember that the threshold sets the *level* at which processing starts or stops, and the attack and release controls set the *time* it takes for the processing to start or stop once the threshold has been crossed.

Most comp/limiters have some type of metering, to show the amount of gain reduction (in dBs) that's occurring. At first, these meters take a bit of getting used to, since their displays 'increase' as the level reduces. For example, if your comp/limiter has an LED meter, the more compressed a signal above threshold becomes, the more LEDs that light up. This is different from a regular LED level meter, which would light up more LEDs as the signal became louder.

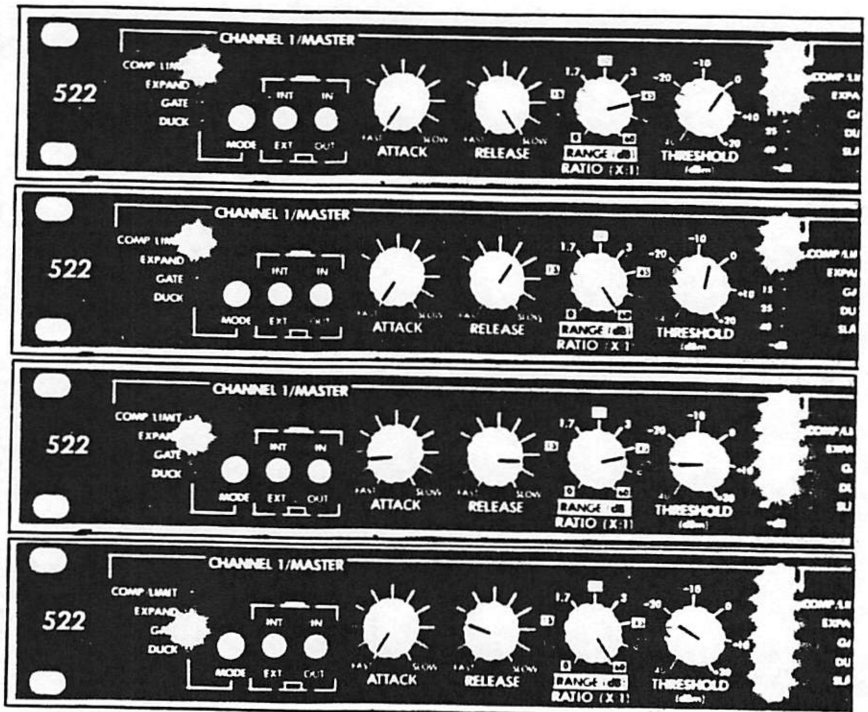
There are some musically useful effects which can be performed by compressing and limiting signals. Compressors, for example, are often used for getting smooth, sustained. bass sounds. By compressing bass notes, attacks are lower, but decays are sustained, since compressors make quiet passages relatively louder.

Here's how to get a sustained bass sound from a comp/limiter. This will also work with a synthesizer or sampler with a bass guitar patch or sample, or with any instrument that has an inherent quick attack and slow decay:

1. Connect the compressor to your mixer, either via a mixer send/receive (insert) jack or an aux send/return (see page 93 if you need more information).
2. Adjust the level of the bass, which is plugged into the mixer, so that the comp/limiter receives an input signal. Your comp/limiter may have an output level control of its own—make sure this is also turned up.
3. Press the comp/limiter's 'bypass' switch, and make sure you get a signal from your bass. If not, check your levels and connections carefully. When everything's okay, engage the compressor (turn off the bypass).
4. Set the attack to 'fast,' and the release to 'slow.' They can be readjusted later, but for now these settings are best, in order to have the compressor respond quickly and continue to respond.
5. Set the ratio control to approximately 6:1. You may wish to lower this to 3:1 or

less, but this is a good place to start, in order to hear the compressor at work.

6. Adjust the threshold control to its lowest level (usually fully counter-clockwise). This is also re-adjustable.
7. Now play a moderately loud note on your bass. An open string will work well, and let it sustain. Your gain reduction meter will light up, as the device compresses the signal, and lowers the note's attack. As the note lowers in input level, the compression will lessen.
8. You should hear the comp/limiter at work—try bypassing the device to compare the dry signal with the compressed signal. The note's attack should sound quieter, with less impact than the dry signal, but the apparent sustain of the note should be greatly increased.
9. Now try raising the threshold level slightly, and set the attack control somewhat slower. When the threshold is raised, the note will have to be louder to be compressed, and with a slower attack, the initial attack of the note will be less compressed, restoring some of the 'punch' and life of the note.
10. When you feel comfortable with these settings, and what effect they induce, try readjusting the ratio and release controls. Take careful note of how things sound different as you readjust the controls—you may find a setting which sounds great with your particular instrument!



Figures 8.23a, 8.23b, 8.23c, 8.23d Various dynamic controller settings, shown with a Symetrix 522 comp/limiter/expander/gate/ducker. From top to bottom we see: 1) A typical 'soft' compressor setting for sustaining bass or controlling a vocal; 2) A 'hard' limiter setting, to prevent overload; 3) A moderate expansion setting, for reducing noise or background sounds; 4) A 'hard' gate setting, for silencing a track instantly. Note that the threshold setting is dependent upon the level of the incoming signal, and that the meter is showing the amount of processing that's taking place.

It takes a lot of practice to feel comfortable using any dynamic processor, since their adjustments are much more subtle than those, let's say, of a digital delay or equalizer. Below is a chart of some various settings you might find useful with different instruments for different effects.

De-Essers And Duckers. Two variations on the theme of compression are *de-essing* and *ducking*. While there are dedicated devices, most comp/limiters can be configured to behave like de-essers and duckers. Let's find out more . . .

A common problem when using a microphone is excessive *sibilance*—'sss,' 'sshhh,' 'tshh,' and other such sounds. Try saying *sensuous slithering snakes solicit solace* and you'll hear sibilance. Now if you try saying the same sentence into a microphone, there's a good chance the sibilance will be overbearing, since many mikes are designed to offer enhanced frequency response at sibilant frequencies, in order to improve articulation in live performance settings. De-essers are designed to reduce the problem by compressing the appropriate frequencies—about 2000 to 6000 Hz—at which the sibilance occurs. Compression is more effective than equalization, by the way, since the greater the level of 'sss' sounds, the greater they're reduced. Using an equalizer alone would reduce the 'sss' sounds by a fixed amount.

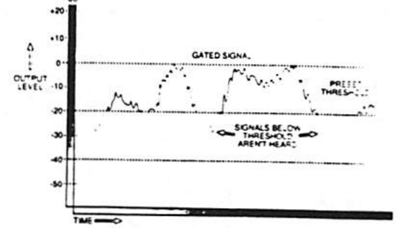
Dedicated de-essers usually have an *amount* control, which adjusts the compression ratio, and a *tune* control, which selects the problem frequencies. When a comp/limiter is used as a de-esser, it's necessary for to 'tell' it which frequencies we want compressed, since otherwise a comp/limiter compresses all frequencies evenly. In order for the comp/limiter to know which frequencies we want compressed, we need to use an equalizer and a special input to the compressor, known as the *side chain* (also called *key* or *control loop*) input. Please see "The Side-Chain" on page 58 for a description of how this system works.

We've all heard radio advertisements where the announcer 'talks over' the background music. During pauses in such ads, we've heard the music get louder, only to become quieter once the announcer resumes. There are two ways to do this. The first is to have the announcer or engineer 'ride gain,' so that the fader levels of the music and announcer are manually adjusted. The other is to use a ducker, which can automatically reduce the gain of one signal—such as music—in relation to another signal—such as an announcer. Almost all current duckers are in fact compressors, with the side-chain input being used to 'duck' the compressor's input signal beneath the level of the side-chain input. This method is also described in "The Side-Chain."

Expanders And Gates. Earlier we described how compressors can reduce a signal's dynamic range. As an example, we mentioned how live music recordings of 80 dB or greater are compressed to fit within an LP record's dynamic range of 60 to 70 dB. Expanders are functionally the opposite of compressors. Whereas compressors reduce the difference in level between quiet and loud passages of music, expanders increase the level between quiet and loud passages. By playing back an LP record through an expander, for example, you can recover the lost dynamic range.

Expanders function with the same sort of controls as are found on compressors, though the threshold control works in an opposite manner to compressors: With an expander, signals are processed when they drop *below* the preset threshold, and signals which are above the threshold are unprocessed. Consequently, it's common to set the threshold control quite high when using an expander.

Musically, expanders have a more specialized function than compressors. As we've described, they can be used to restore or further enhance an instrument's—or group of instruments'—dynamic range. In some settings, this can create a more 'alive' sound, though the trade-off is that quiet passages often can be made too quiet, and get 'lost' in noise. Expanders can also be used to reduce noise—such as tape hiss—



Figures 8.24a, 8.24b The effect of an expander and a gate on a signal such as that of Figure 8.21a.

since noise is usually low level, and expanders make low level signals even lower.

Typically, expanders use ratio settings of 1:3 or less. The ratio is also backwards, compared to compressors: A 1:3 ratio would mean that for every 1 dB of input gain, the output gain would increase by 3 dB. Similarly, for every 1 dB of input decrease, the output would decrease by 3 dB.

Gates are a type of expander, and they're more commonly used than the 'regular' expanders just described. Gates, also known as *noise gates*, have a simple-sounding task: They are either 'open' or 'shut,' depending upon the input's signal level. When input signal levels exceed the gate's threshold, the gate opens—signals pass through to the gate's output without any processing. When signal levels are below the gate's threshold, the gate 'shuts'—the signal doesn't pass, and we hear nothing from the output of the gate. In fact, gates are expanders with extremely high ratios of 1:10 or more. Many expanders can function as gates, much as compressors function as limiters.

Gates have all kinds of musically and sonically useful roles. In addition to singing, for instance, a mike can pick up all sorts of unappetizing sounds, such as lip smackings, breath noises, throat clearings between phrases, and so forth. With a gate, the threshold control can be set so that any singing passes through without being 'gated,' and so that all relatively quiet sounds are gated. If the controls are set properly, the result is a nice, clean vocal track, with singing during the vocal phrases, and silence during the pauses. Similarly, gates can help rectify crosstalk between musical phrases on a track, or any other low-level intrusive noises.

Just as with comp/limiters and expanders, gates often have attack and release controls, which determine how quickly the gate 'kicks in,' or closes, and how long it takes for it to open up. Some gates have just a threshold, attack, and release control, with a non-adjustable ratio. Multi-channel gates are popular: Some have as many as four

The Side-Chain

The side-chain—a component of almost all rack-mount dynamic processors—plays a very useful though often misunderstood role. To learn about its utility, let's look at a general model of how dynamic processors work....

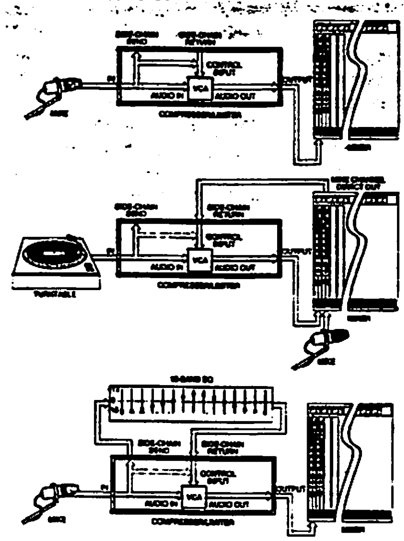
In a typical dynamic processor, an input signal enters the device, is processed by the VCA (voltage controlled amplifier), and leaves as a processed output signal. The VCA is itself controlled by whatever signal is present at the device's side-chain. Normally the side-chain signal is the same as the input, so that the processing—compression, gating, whatever—occurs in response to changes in the input's level. The processing is then applied to the input signal. Figure 8.25a shows a block diagram of a normal processing arrangement, with a microphone being processed.

What if we were able to access that side-chain? If we could, we would be able to let signals *other* than the device's input level control the processing. In that way, the signal could be processed by something completely different. It so happens that most rack-mount dynamic processors do allow us to access the side-chain, with a side-chain—also called a *key* or *control loop*—input. Why is this useful to us?

Take a look at Figure 8.25b. What's pictured here is a typical ducking situation. We have a turntable as the input to the compressor, and a microphone connected to the side-chain input. Normally—without anything connected to the side-chain—the turntable would 'compress itself,' so that as its signal became louder or quieter, its level would be reduced or increased. In the situation pictured, however, as the level of the microphone increases above threshold, the turntable's signal is compressed and lowered. As the level of the mike decreases, the turntable's signal is raised. In other words, the mike is ducking the turntable—a DJ's dream, and a perfect example of how an external signal at the processor's side-chain can control the input signal.

De-essing is another popular use of a compressor's side-chain. Here's how it works: Take a mike signal (or a mike track from tape), and split it so that one signal goes to the compressor's input, and the other goes to an EQ (Figure 8.25c). At the EQ, boost the troublesome sibilant frequencies—usually between 2000 and 6000 Hz while cutting the non-sibilant frequencies. Then take the output of the EQ and run it into the compressor's side-chain. The result? The sibilant frequencies are controlling the VCA, and they'll reach the threshold level before the non-sibilant frequencies. In turn, the VCA will induce *frequency-dependent* compression, and compress the problem frequencies of the input signal before it compresses the regular frequencies. Lo and behold, a de-esser from a compressor.

When using the side-chain, keep in mind that it's the input to the side-chain which crosses the threshold and controls the VCA. Consequently, the levels of the regular input have no effect on the device's threshold.



Figures 8.25a, 8.25b, 8.25c Normally, a dynamic controller's VCA is controlled by whatever signal is entering the device (top). With the side-chain, however, a comp/limiter can behave as a ducker (middle), or even as a de-esser (bottom). The side-chain send is useful for any frequency-dependent processing.

Side-chaining a gate can be a really effective practice. One technique is frequency-dependent gating. For example, let's say you were miking an acoustic drum kit, and everything sounded great except for a squeaky kick drum pedal, which was being picked up by the hi-hat mike. The problem could be reduced in this manner: Split the hi-hat mike's output, so that one signal went into a gate, and the other to a parametric EQ. At the EQ, *cut* the offending squeak's frequency—which might be 1500 Hz or so—then run the output of the EQ into the side-chain input of the gate. Consequently, if the threshold is set properly, the gate opens for all frequencies except the squeak: If the pedal is moved just on its own, it won't generate enough level through the EQ and side-chain input to open the gate. Of course, when the hi-hat is being played, the gate is open, but the squeak should be masked by the volume of the hi-hat.

Frequency-dependent gating is very common in top studios—so much so that some gates made by Drawmer and other companies include a built-in EQ, which can be sent to the side-chain with a switch. Note that when you use an EQ in the side-chain, you don't equalize the input signal, you simply determine which frequencies first cross the threshold of the VCA. Also note that some processors include a side-chain *send*, which splits the processor's main input signal, so that you can send a signal to an EQ or other device.

The other applications for the side-chain are too numerous to mention, and no doubt there are many applications waiting to be discovered. Use your imagination! Here's one last gate application which might encourage you to discover your own side-chain tricks: On Kate Bush's *Hounds Of Love* album [EMI ST-17171, 1985], there's a cut called "Waking The Witch," where Bush's voice is interrupted continuously and rhythmically while she's singing, as if someone were quickly switching off and on the mute button for her vocal track. It carries on for some time, so it soon becomes clear that no engineer could have the timing, speed, or endurance to be hitting a switch. How is that effect achieved?

One way to create a rhythmic on/off for a track would be to use an elaborate automated mixing system, though there's a simpler, less expensive way, by creating a *trigger* for the gate: Run the vocal track through a gate, and adjust the gate threshold so that the vocal is at too low a level to open the gate. Then take a percussive source of rhythm, such as a kick drum from a drum machine (it's not necessary that the source be heard). Plug the drum into the side-chain, and set the threshold so that the gate opens whenever the drum hits, and closes when the drum is silent. If a rapid attack and release are chosen, you'll hear the vocal cut off and on, as the drum triggers the gate open each time it hits.

independent gates. And some comp/limiters have built-in gates, such as the dbx 166 and the Symetrix 525. With these comp/limiter/gates, there's usually just a single threshold control for the gate, with preset ratio, attack, and release settings.

In the early 1980s, British producer Hugh Padgham popularized a very distinctive drum sound with Phil Collins and other artists. His technique is to take a highly reverberant drum sound, such as a snare, and gate it. By adjusting the gate's attack, the reverberant snare can be heard for half a second or so, until it suddenly disappears. When hit on the following beat, the snare sounds as if it's exploding out of silence.

This gated drum sound is so effective because it's performing a bit of a sonic trick on our ears: When we hear a lot of reverb, we're conditioned to hearing a long decay, perhaps two seconds or more, accompanying it. In addition, a lot of reverb can suggest a very 'big' sound, as if we're listening in an arena. So when we hear that snare drum suddenly cut off, the result is very dramatic—it's as if briefly, someone opened a soundproof door to a giant arena with a giant snare, and then suddenly slammed the door shut!

If you record live drums or own a drum machine with an individual output for the snare, it's very easy to create this effect: Simply run the snare channel's output into a reverb, then route the output of the reverb into the gate. By adjusting the threshold, attack, and release controls in various ways, you can create different gated reverb sounds, with varying decay rates and intensities.

As with other dynamic processors, many gates have a side-chain input—once again, please refer to "The Side-Chain."

Other Signal Processors

Equalizers, spatial effects (delays and reverbs), and dynamic processors are the most common and generally useful signal processors. There are many other processing devices on the market, however—some popular, some obscure. As digital processing continues to allow more features for less money, we can expect to see more of the following effects.

Pitch Transposers. Also known as a *Harmonizer*, for the first such device marketed by Eventide, a pitch transposer does as its name suggests: It can transpose the pitch of a note or group of notes. The degree of transposition is selectable on the front panel of the device, and some transposers allow you to store preset settings, for instant recall.

Pitch transposers are usually digital devices, capable of storing a sound as a group of binary numbers, manipulating that sound, and spitting it back out as the transposed signal. On most units, transposition can be as little as a musical quarter-tone or less, or as much as two octaves.

Performance artist Laurie Anderson has used pitch transposers extensively both on her violin and with her voice. A common effect of Anderson's is to pitch her voice down an octave or so, so that her voice sounds like a male voice. Aside from these types of special effects, pitch transposers can be useful for 'thickening' the sound of a voice or an instrument. The method is to mix the dry signal with a slightly off-pitch signal: The combined signal sounds fuller than the original dry signal. In addition, with the help of a pitch transposers you can correct tape tracks that were recorded at wrong speeds, or even correct a slightly off-pitch note.

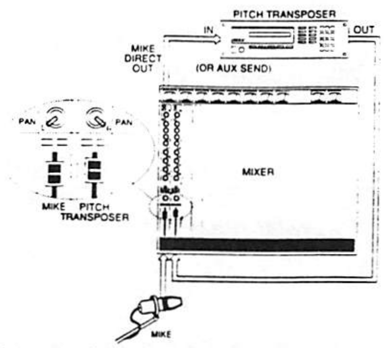


Figure 8.26 Thickening with a pitch transposer.

Exciters And Enhancers. In the late '70s, a mysterious device known as the Aphex Aural Exciter began to show up at some Los Angeles studios. It was for sale, but in order to use one for a commercial recording, it was necessary to sign a licensing agreement and pay royalties to Aphex. If that wasn't enough to generate talk, a lot of people felt that the Exciter could turn a good-sounding recording into a great-sounding recording. In fact, once some engineers heard their best recordings processed by an Exciter, they refused to work without one. Today, there are still some engineers who swear by their Exciters, recording the effect on each individual track as well as the complete mix. What is this *wunderbox*?

Essentially, any signal that's been processed by an Exciter sounds brighter, and perhaps more 'alive.' Typically, the Exciter is connected between the console and the 2-track during mixdown. By adjusting some controls on the front of the device, you can add the desired amount of effect. And what is the effect? In essence, when you 'excite' a signal, you are adding musically-melodic harmonics—the device is adding clean-sounding, third harmonics to whatever music is being excited. The bright-sounding result is different from an equalized signal, since the harmonics change with the music, and are not an across-the-board boost of certain frequencies.

An exciter isn't for everyone, however. Some people find them harsh sounding, particularly if used at a high setting. Other people find that at first the excited signal sounds great, only later to experience 'listener fatigue'—their ears simply get tired of hearing the bright, excited sound.

In 1985, a new type of signal enhancer appeared on the market by BBE, a division of Barcus Berry (a company which specializes in contact transducer/pickups for acoustic instruments). The BBE enhancer sounds at first a bit like an exciter, adding a dramatic amount of 'life' to the music. Unlike some exciters, however, few people report listening fatigue with the BBE. In fact, the BBE processor seems to make it *easier* for people to listen to music for extended periods of time, while making the signal sound bright and alive.

The BBE system does not involve harmonics. Instead, it works by performing *phase correction*. Back on page 3, we learned that when two identical sound waves shift slightly out-of-phase with each other, there's some phase cancellation, and some frequencies are lost. The BBE works on the principal that most audio signals contain slightly out-of-phase signals, and seeks to correct those phase relationships so that the sound is restored to its original in-phase sound. A BBE processor also contains a low frequency enhancement control, which helps to add more definition and punch to the bass.

Many engineers greet these enhancers with skepticism, partly because it's a bit of an offense that some 'black box' can improve upon the sound of their mix; others just dislike their sound. Nonetheless, other engineers praise enhancers as important studio tools. From this perspective, the process of phase correction—rather than harmonic addition—may be the most useful tool. But these are just opinions: As with any processor, the ultimate test must be your own ears and judicious application.

Vocoders. One of the most distinctive effects is that of a *vocoder*, which was originally developed in World War II to encode messages. If you've heard the sound of musical instruments 'talking,' or of voices sounding like instruments, likely you've heard a vocoder. The concept is fairly straightforward: A mike is plugged into a device which contains analyzers. These analyzers divide the vocal into various frequency bands. Each band's volume and harmonics can be controlled by an external signal, such as a keyboard or guitar. In this way, whenever the external signal is played, the corresponding vocal bands take on that signal's character, and cause the vocals to sound as if they are articulating the instrument. The effect is similar to placing a small headphone element or speaker in your mouth and moving your lips to articulate the sound coming

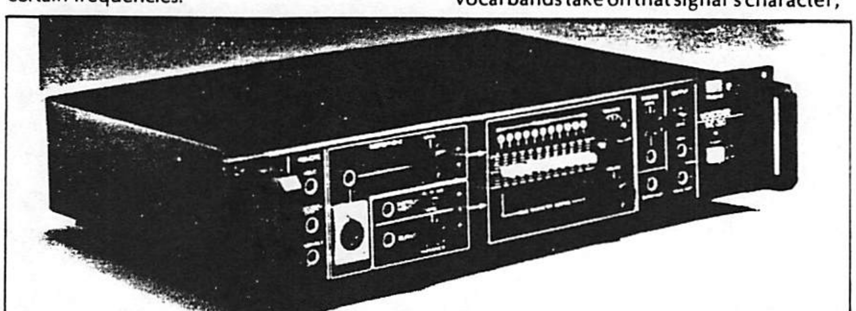


Figure 8.27 The Roland SVC-350 vocoder has 11 bands of processing.

from the speaker.

Laurie Anderson, Stevie Wonder, and Black Uhuru are some of the artists who have popularized vocoders made by Korg, Sennheiser, and Roland.

Noise Reduction

For many studios, *noise reduction* is the most important form of signal processing. There are some professional studios, such as those with digital tape recorders or recording with professional machines at 30 ips, who have no need for noise reduction. For most of us, however, noise reduction is a fact of life.

There are two main categories of noise reduction. The simplest, and least common category, is known as *single-ended* noise reduction. Symetrix, Rocktron, and other companies offer these as outboard devices, designed to reduce noise from any source, such as a tape recorder, mixer, or whatever. Unlike a gate, which shuts off a signal's flow to eliminate noise, these single-ended processors are more like frequency-sensitive expanders—reducing the volume of low-level high frequency noise, while leaving the rest of the signal unprocessed. Unfortunately, they also reduce some of the high frequency information to varying degrees.

When most people think of noise reduction, they have in mind the most common category: *tape* (also known as *encode/decode, two-stage,* or *compander*) noise reduction, such as dbx or Dolby C. For the balance of this section, when we speak of noise reduction, we're talking about this category.

Virtually all gear in any studio produces some level of noise. In analog (non-digital) studios, tape hiss is the most problematic noise source, assuming everything else is working okay and that there are no external noises being induced into the system. As we learned back on page 5, when tape is unrecorded, its oxide particles are at random. If we just listen to blank tape, without noise reduction, we hear a hiss: That's the sound of random particles at rest, and what we're hearing is a rather broad and even range of frequencies, similar to *white noise* (which is the sound of all frequencies at the same level). After we make a recording, we can still hear some unoriented particles, and that's the sound of *residual*, or 'left over,' tape hiss.

The amount of tape hiss is dependent upon many things: The type of tape used, the quality of the recorder's electronics, and so forth. All other things equal, however, tape hiss increases as tape speed and track width is reduced. For example, a 1/4" 8-track tape recorder operating at 7½ ips will suffer much more tape hiss than a 1" 8-track operating at 30 ips. Most 4-track cassette recorder/mixers are unusable without noise reduction.

Referring back to the definitions of signal-to-noise on page 21, few people would want to use a recorder that had a S/N ratio much worse than 50 dB, without noise reduction. Tape recorders operating at 30 ips—including 1/2" 8-track, 1" 16-track, and 2" 16- or 24-track—can sound quite quiet without noise reduction, offering a S/N ratio of 60 to 70 dB. Below these thresholds, however, noise reduction is important if not essential. Noise reduction is either built-in, as with all ministudios and many open-reel machines, or available as an outboard processor, located between the mixing console and the tape recorder.

The most common noise reduction systems are made by dbx (Type I and II) and Dolby (A, B, and C). All of these systems share a few things in common. For one, they require that the recording be *encoded* with noise reduction during the recording; this encoded signal is later *decoded* by the noise reduction system during playback. Unlike single-ended noise reduction, which can be applied to any source being played back, this encode/decode process is a necessary part of all systems made by dbx and Dolby—encoded signals played back without decoding sound overly bright (as with Dolby) or virtually unlistenable (as with dbx).

As part of the encode/decode process, both systems use compression during recording and expansion during playback; that's why they're called 'companders.' Remember, compression reduces dynamic range, and expansion increases dynamic range.

Basic Theory. Turn on a water faucet, and listen to the flow of the water. If there's no other loud sound in the room, the water sound should be very noticeable—in much the same way as tape hiss is noticeable in the absence of any music.

With the water still running, play your stereo quite loud. Even though the water is still running, the volume of the stereo should *mask* the sound of the water—so that all you hear is the music. At this point, you have a high signal (music) to noise (water) ratio. Finally, adjust the volume of the stereo up and down, so that you are able to occasionally discern the sound of the running water.

This model is a perfect example of the effect of noise during a recording. When the music is loud, the tape hiss is masked; when the music drops in level, the tape hiss becomes audible. If this suggests to you that one is more likely to hear tape hiss with dynamically varying music (such as chamber music) than with consistently loud music (such as 'heavy metal'), then you understand the nature of noise and masking. For now, tuck this bit of information away, and let's look at companding.

Tape noise is a constant. Without noise reduction, its level stays the same, no matter how loud or quiet the music. Noise reduction's purpose is to reduce our perception of that constant, and the basic idea is as follows: The encoding process *compresses* a signal before it's recorded on tape, reducing its dynamic range. Remember that compression makes loud musical passages quieter, and quiet passages louder.

That's the encoding process. We need to decode, and here's why: A compressed signal has a reduced dynamic range, causing everything to sound much less lifelike. If we compress a symphony orchestra at a 2:1 ratio, its dynamic range is reduced by 50%. Listening to an encoded tape without decoding is not a musical treat. The problem is solved, however, by *expanding* the signal during the decoding process.

Expanding during decoding does more than just restore the dynamic range: It's *the key* to reducing the noise. Take a look at Figure 8.28: We start with the original, unencoded music, with a dynamic range of about 90 dB in this case. Next, the music is encoded by compression, and then recorded onto tape. For our example, the compression ratio is 2:1. As we can see in our example, the tape hiss is sitting at a level of -45 dB, and the compressed music is recorded at levels from -40 to +5 dB.

When the tape is decoded during playback by expansion, everything is expanded at a 1:2 ratio, including the tape hiss. The result is that loud music passages are made twice as loud, and quiet passages are made twice as quiet. The final step shows what we hear: The music is restored to its original 90 dB dynamic range, and the tape hiss is expanded downward from -45 dB to a level of -90 dB—which is barely audible!

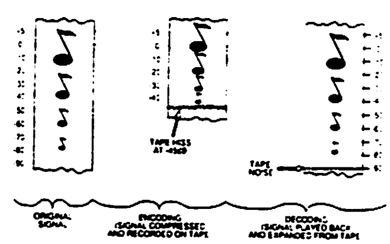


Figure 8.28 Compression and expansion: The basic principle of 'compansion' noise reduction.

Bear in mind that with any encode/decode tape noise reduction system, only tape hiss and other noises caused by the recording process are reduced. Any noise which is recorded with the original signal—such as a buzz from a guitar amp, or hiss from the mixer—will remain the same and not be reduced!

That's the theory. In practice, there a few different approaches . . .

dbx. There are two types of dbx noise reduction. Type I works best with high-speed tape recorders that have a relatively flat frequency response across the audio spectrum from 20 Hz to 20 kHz. Type II is most common in broadcast and cassette applications, since it's optimized for use with slower-speed decks. Either can be calibrated for +4 dBm or -10 dBV levels.

For the most part, dbx operates as the compander system pictured in Figure 8.28. There is one major difference, however. During the encode stage, dbx boosts the high frequencies with a simple equalizer circuit: This boost is known as *pre-emphasis.* During decoding, those frequencies are returned to normal with another EQ circuit, in a process known as *de-emphasis.* By lowering all the high frequencies during playback, the benefit is a further decrease in noise, since tape hiss is largely high frequency information. All in all, dbx can reduce tape hiss by a lot—as much as 30dB.

Dolby. There are three common types of Dolby noise reduction: Dolby A, B, and C (Dolby SR is a less common type, which we'll look at in a moment). Dolby A is the most effective, reducing noise by 15 to 25 dB. Dolby C follows, with an average 20 dB reduction, followed by Dolby B, which reduces noise by about 10 dB.

Like dbx, and as shown in Figure 8.28, all three Dolbys are companding systems. Here's one major difference between dbx and Dolby, however: The Dolby systems

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switch off when the levels are consistently loud enough to mask the residual tape hiss. This contrasts with dbx, which reduces the amount of noise reduction, but doesn't switch off, and represents a philosophical difference. The Dolby people feel that it's not necessary to apply processing 100% of the time, but only when it's necessary. Consequently, Dolby circuits have level detectors which disengage the encoding or decoding when the overall signal levels are loud enough.

Another difference between dbx and Dolby is that dbx applies the same companding to the entire frequency band, to reduce noise at all frequencies; Dolby systems don't offer the same across-the-band reduction. For example, Dolby B only works from 1000 Hz and up; below 1000 Hz there's no Dolby B processing. This is another philosophical difference: The Dolby people feel that since tape hiss primarily happens at higher frequencies, that's where the processing should take place, whereas the dbx people feel that noise at any frequency is of consequence. Dolby does agree, to some degree, with dbx on this issue. For example, Dolby C operates on two independent bands, from 400 Hz to 1000 Hz, and from 1000 Hz on up. And Dolby A, the most expensive and effective system, operates on four different bands: from 80 Hz on down; from 80 to 3000 Hz; from 3000 to 9000 Hz; and from 9000 Hz on up.

Other Systems. The most powerful noise reduction system of all is a new type of Dolby, which operates on much more complicated and elaborate principles than the ones described above. Known as Dolby SR, for *spectral recording*, it's a multi-band system that is capable of providing performance that rivals the world's most expensive digital tape recorders. Dolby is guarding the process very carefully, however, and as good as it is, inexpensive digital recording may beat SR into the home studio: SR costs over $1,000 per channel.

In Europe, noise reduction systems made by ANT/Telefunken, known as Telcom, are quite popular. One system resembles Dolby A, and another aims to compete with dbx and Dolby SR. Just like Dolby A and SR, however, Telcom is not commonly found in the home studio.

Growing in popularity is something known as Dolby HX Pro. This is *not* a noise reduction system, *per se*. Rather, it's a built-in circuit—developed by Bang & Olufson and Dolby—which improves the headroom of recording circuits, so that there's less tape saturation (covered on page 20). The result is that higher levels can be recorded with less loss of high frequencies. HX Pro is often found in conjunction with Dolby C or other noise reduction systems.

Which System Is Best? Many professionals would agree that Dolby SR is the best noise reduction made. For most people, however, its cost is prohibitive.

In home recording, there are three common noise reduction systems: dbx Type II, Dolby C, and Dolby B. Dolby B is the least effective of these three, and is the only noise reduction for a few of the least expensive ministudios. For the rest of the market, the debate of dbx vs. Dolby C is hotly contested. Tascam, Yamaha, and others would have you believe that dbx is best; Fostex, AMR, Audio-Technica, and others advocate Dolby C. These opposing camps exist largely thanks to effective marketing by both dbx and Dolby, though there are true believers on both sides of the fence.

Here are the facts: dbx gives you more headroom than Dolby C, so that you can record 'into the red' on your meters, with less fear of tape saturation. As well, dbx does reduce more noise than Dolby C. In fact, if you just listen at a high level to blank tape with dbx engaged, you'll hear practically no noise; with Dolby C you'll still hear a tiny bit of hiss—though a *lot* less than with Dolby B. But most of us spend little time listening to blank tape—how do dbx and Dolby C sound with music?

Some people claim Dolby C sounds best because it doesn't process all frequencies at once: It has two separate bands which turn on separately when levels get high enough. Since dbx works all the time on all frequencies at once, some feel you can hear it working from time to time. For example, with dbx, when just low notes are played back, it's possible occasionally to hear the hiss expanding upward—this is what people refer to when they claim that dbx 'pumps.'

There are two other categories where Dolby C may be preferable to dbx. When syncing drum machines and other devices to tape via a sync code (as we'll examine on page 112), in most cases it's necessary to record the code without dbx, though it's okay to record it with Dolby C. The reason is that the processing of the dbx circuit can distort the code. Tascam and other manufacturers have conquered this problem on some ministudios by installing a 'dbx off' switch for one of the tracks, or by offering a 'sync input' and 'sync output'—which is an input and output to one track (usually track 4 on 4-tracks) which bypasses the dbx circuit.

Finally, another quirk of dbx is related to headbump: You may recall, back in Figure 5.4, we learned that the design of tape heads results in a slight increase of frequencies around the 120 Hz range. A noise reduction system can worsen this headbump in the following way: One potential problem with *all* companding noise reduction systems is that any frequency deviations—increases or decreases in levels from flat response—can become exaggerated. When tracks are bounced, or ping-ponged together, a companding noise reduction system can further exaggerate deviant frequencies. Since dbx works on all frequencies, including headbump frequencies, if tracks are bounced together without switching off the dbx, the expansion process can cause the headbump to audibly double. Musically, this can make a bass guitar sound 'boomy' and undefined. Dolby C doesn't work at frequencies below 400 Hz, so bouncing with Dolby C engaged does not worsen headbump problems.

So far, it may sound as if Dolby C is the clear winner over dbx: This is *not* necessarily the case! Some users enjoy the extra degree of true noise reduction that dbx offers over Dolby C. And Tascam, Yamaha, and other manufacturers who incorporate dbx in their gear, have effectively addressed the problems we've described by including 'dbx defeat' switches on ministudios. These defeat switches can work on a single track—for sync code—or they can work on all tracks—for bouncing. If you're shopping for a recorder/mixer with dbx, make sure you find one with these defeat options. If you do, the problems we've described will be alleviated, and you can count on a noise- and trouble-free recording.

If you own an independent open-reel machine without noise reduction, such as any of the Tascam or Otari machines, your outboard noise reduction choices are dbx, Dolby A, Dolby B, or Dolby SR. Dolby no longer licenses the manufacture of outboard Dolby C processors.

In the last analysis, your ears must once again make the decision—whatever your choice, however, any quality system will help you make clean, quiet tape recordings.

One final note about noise reduction, which addresses a problem that many of us have encountered. The old adage about Dolby B is that "It should be switched off—it reduces high frequencies!" Certainly many of us have defeated the Dolby B switch when playing back tapes, to cure a muffled sound. Dolby B does not reduce high frequencies, however, unless it is improperly calibrated. And the fact is, very few cassette decks, multi-track or not, are properly calibrated by the time they reach your living room from overseas. The solution: Have a technician properly align and calibrate your tape machine! This applies with any tape deck, and with any noise reduction system. Dolby C and dbx are more tolerant of miscalibration than Dolby B, but nonetheless, calibration will greatly improve your tape deck's performance.

* * * *

With equalizers, delays, reverbs, comp/limiters, gates, noise reduction, and more, there's a lot of signal processing gear from which to choose. Even the best studios are not completely equipped with every type of signal processor on the market, and one of your tasks in setting up a studio will be to decide what processing will serve you best.

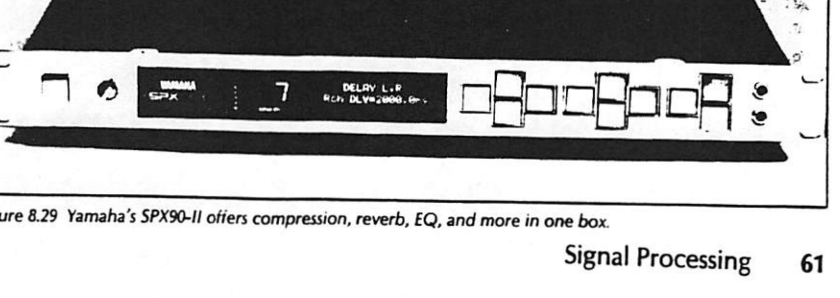


Figure 8.29 Yamaha's SPX90-II offers compression, reverb, EQ, and more in one box.

Fortunately, as we noted earlier, a growing trend is the incorporation of many effects into one box. The Yamaha SPX90-II is a prime example, offering reverb, delay, pitch transposition, compression, and more, in one relatively affordable package. In a similar manner, we can expect to see effects other than just EQ incorporated into more mixing consoles.

All of this portends good news for both personal and professional recording studios, offering us greater creative options. Though signal processing can do little to improve a poor performance, it can make a great performance sound even better. □

CHAPTER 9: MICROPHONES

In the preceding chapter we learned how signal processing for the engineer is analogous to lighting for the photographer, since both allow moods and atmosphere to be created and expressed. If that holds true, then microphones are our lenses, and determine how we audibly 'view' our acoustic subjects. And just as different photographers will approach the same subject with different lenses, veteran engineers are more opinionated about microphones and miking techniques than perhaps any other topic.

With the advent of affordable digital drum machines, and with sampling keyboards capable of remarkable piano, string, and other lifelike sounds, the role of microphones in the average home and professional studio has diminished greatly. Whereas an arsenal of six to 15 or more mikes may be necessary to record acoustic drum kits and pianos, many home studios manage fine with one or two quality mikes, for vocal or guitar tracks—or for recording samples!

While the overall role of mikes may be diminished, however, perfectly recorded digital drum sounds and other high quality samples challenge us to make mike recordings of equal quality. For those of us with just one or two mikes, our choices and techniques are as important as ever.

If you find yourself wanting to learn more about mikes and miking techniques than this book can allow space for, *The Microphone Handbook*, by John Eargle, is highly recommended. The world of microphones encompasses many phases of science, art, and even some witchcraft, and deserves a book such as Eargle's.

Before we get into the details, here are a couple of things to keep in mind: Microphones can be *high* or *low impedance* ('Z'), and can be *balanced* or *unbalanced*. As we learned back on pages 22 and 23, low-Z (600 ohms or less) balanced mikes are preferable to high-Z (2000 ohms or higher) unbalanced mikes, since they are less susceptible to noise and can have cable runs of greater than 20 feet without high frequency losses.

If you already own a high-Z unbalanced mike, or low-Z balanced mike, the following information will apply to you. Remember though, even if your mixer or ministudio only has 1/4" unbalanced mike inputs—as do most ministudios—low-Z balanced mikes can be used with a transformer, and will offer better performance.

Microphone Types

In Chapter Two we learned how a microphone is a transducer, capable of turning sound energy into electrical energy. While there are many mikes on the market, almost all fall into one of three categories: *dynamic*, *condenser*, and *ribbon*.

Dynamic, or Moving Coil Mikes. Referring back to Figure 2.2, we can see a microphone with a wire coil attached to its diaphragm. As sound strikes the diaphragm, it moves—causing the coil to move within a fixed magnet and in turn generating electrical signals. This is a description of a dynamic, or *moving coil*, microphone.

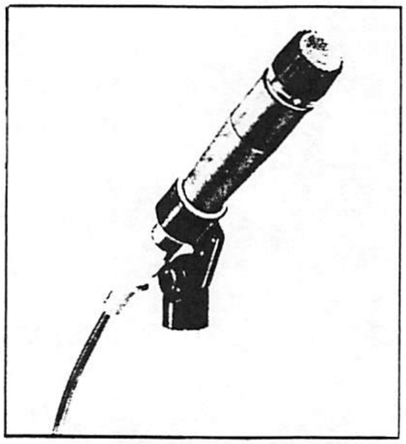


Figure 9.1 The SM-57 is very similar to the '58,' but lacks the built-in pop filter—it's a popular choice for drums.

Dynamic mikes are very common in live performance, since they're traditionally inexpensive compared to condenser and ribbon mikes, and their design makes them rather indestructible; physical trauma can lessen their quality, but they'll usually continue to make sound.

Another benefit of a typical dynamic mike is its ability to resist overloading. That is, its diaphragm is resistant to distortion, since it's relatively stiff—even the loudest guitar amplifier or kick drum can be picked up without distortion from most dynamic mikes.

All this ruggedness and resistance to distortion has a price: Many dynamic mikes have a relatively uneven frequency response when new, and much worse after abuse. Many also have a poor *transient response*. A *transient* is a relatively high level signal with a sharp attack and decay, such as the instantaneous attack of a guitar string or a stick on a snare drum. In order to mike these kind of transients accurately—and to keep them sounding bright and punchy—a diaphragm has to be able to move quickly. The problem with many dynamic mikes is that their diaphragms are too slow to do this, since the attached coil adds a relatively large amount of mass.

There are exceptions to the rule, however: The Beyer M201, for example, is a dynamic mike which has been designed to offer superior transient and high-end frequency response. And there may be many circumstances which don't require great transient or high frequency response, such as miking a guitar amp. Similarly, there are some miking situations which simply demand ruggedness or resistance to distortion, such as miking a kick drum.

Condenser Mikes. Condenser microphones don't use a moving coil and magnet, as we described for dynamic mikes. Rather, they use a *capacitor* to transduce electricity from sound.

Here's how they work: A capacitor is an electronic device with a voltage potential between electrically-charged 'plates' in close proximity, one positive in charge, the

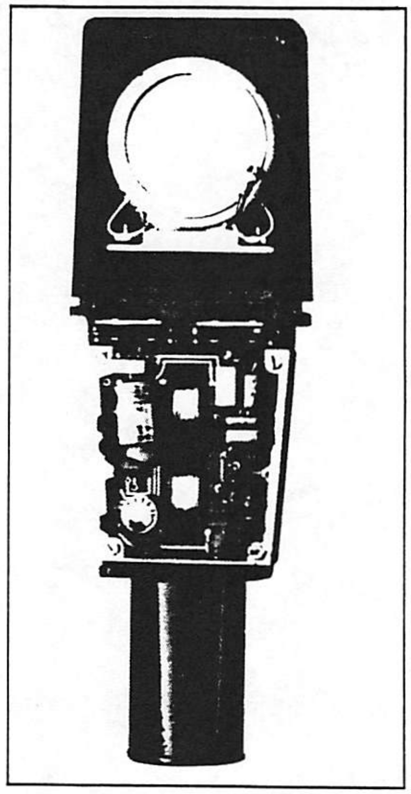


Figure 9.2 A cutaway view of an AKG C414 condenser microphone. Like many finer condenser mikes, the C414 has a large diaphragm, and requires 48 volts of external phantom power. Beneath the diaphragm we see the mike's preamp. Most mikes, incidentally, can be repaired if damaged by a sudden blast of air, such as from a kick drum: When the diaphragm is replaced, they're almost as good as new.

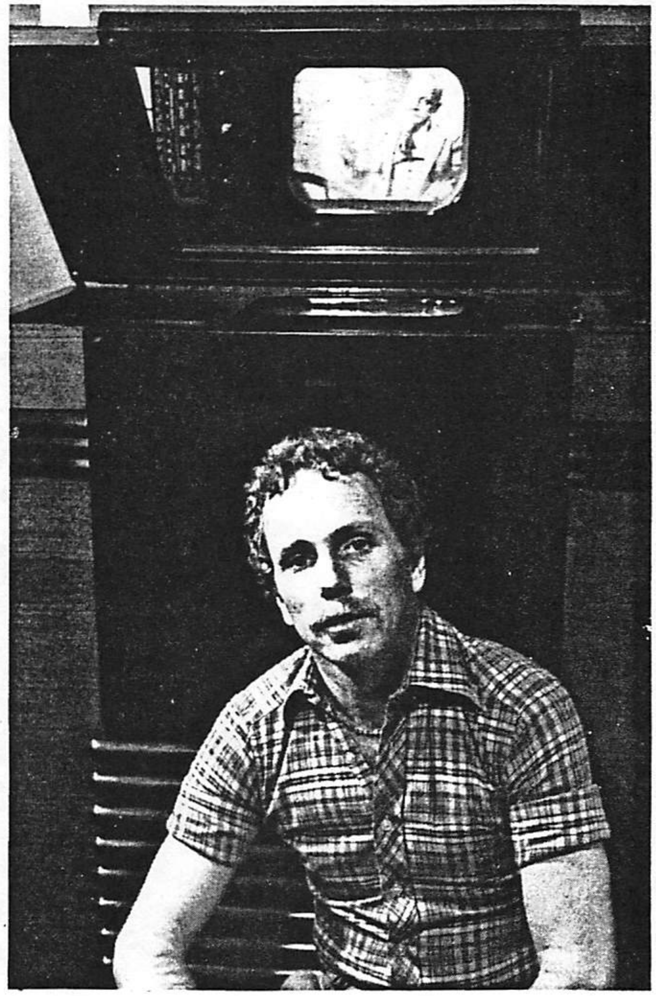
COLLECTING VINTAGE TELEVISION SETS

KENT WARNER

When I lived in New York City, back in the late 1950's, there was a small area of second-hand radio and TV shops in lower Manhattan known as "Radio Row." Within the battered confines of these decrepit stores one could find masses of vintage radio and television sets of every shape, size, and description. Because I was quite young at the time, and did not have much money, I couldn't even consider buying any of these "talking-seeing" pieces of furniture. But I was curious nonetheless, and after I moved to California in 1962, I became an active collector of radio and television sets.

I returned to visit New York in the winter of 1975, and of course I headed directly (cash in hand) for that packed, electronic paradise I remembered from the 1950's. Much to my shock and disappointment, however, the entire area had been cleared and replaced by the 110-story World Trade Center. I asked around, but no one seemed to know or care about what had become of all those old radio and TV store owners, or their merchandise. It was all gone—just disappeared. Do you suppose the World Trade Center is delicately perched on an RCA TT5 or a McMurdo Silver chassis, or a Brunswick Panatrope? Whatever the case, I realized *now is the time. We must preserve television's past.*

Contrary to a popular belief, television is not a product of post–World War II American technology. The combined efforts of many minds, working in many countries, over many years, created television



Kent Warner poses with his prized 1939 RCA TRK-12 receiver.

On January 25, 1949, the first Emmy Awards are presented and KTLA in Los Angeles is chosen best station. Its top show, Mike Stokey's *Pantomime Quiz*, wins as TV's most popular program.

The Spade Cooley Show is one of KTLA's earliest successes. It attracts such guests as Frank Sinatra, Jerry Lewis, and Sarah Vaughan and remains on the air until 1956. In 1961, Cooley is sentenced to life in prison for killing his wife. In 1969, just prior to his parole, he performs for the first time in eight years. Backstage, after the show, Spade Cooley dies from a heart attack.

as we know it today. In 1847, in England, Frederick Bakewell was experimenting with sending line drawings by wire. In Germany, Paul Nipkow was granted a patent for sending moving pictures by wire in 1884. He employed a mechanical, or scanning-disk system that was further developed by John Baird in England in 1925, and simultaneously in America by C. Francis Jenkins and the Bell Telephone Laboratories. Early in the 1930's, scanning-disk television was developed to its fullest potential, but it soon became evident that its flickering picture and low definition would not be acceptable to a public that was then accustomed to the better quality of motion picture film.

It was the work of two Americans, Philo T. Farnsworth and Dr. Vladimir Zworykin, that finally enabled television to come of age. Their television tube patents provided the basic elements for the wholly electronic system that we use today. By 1936 the scanning-disk system was outdated and fully electronic television was being experimentally broadcast in England, Germany, and the United States.

By late 1938 and 1939, many American radio manufacturers were introducing television sets to the public with major campaigns to encourage buyers. Sales were slow. Receivers were expensive and there was minimum daily programming available. There were only twenty-one licensed television stations in the United States in 1941, when World War II temporarily halted television's advance. But after the war, television quickly regained momentum, and by the end of 1948 there were over one million television sets in the United States.

Sadly, little remains of those early television receivers. New developments in electronic technology rendered them outdated, and people were quick to discard their old friends for new models that boasted increased dependability and larger screens. Electronically gutted, the superior cabinetry of those early sets still survives here and there and examples of it can sometimes be found hiding in used-furniture stores disguised as record cupboards and liquor cabinets.

One of the first relics of progress that I was able to save from the junk pile was a 1948 Pilot-Radio Candid TV receiver. My discovery was still in its original shipping carton and sported a magnifying bubble over its three-inch picture tube. This primitive-looking receiver sparked my interest in early television and marked the beginning of a fascinating hobby.

I soon learned that there were some publications on television from the 1930's and 1940's that could be found by those who were willing to plough through dusty bookstores with perseverance. A little such background material is very helpful in revealing the differences between a rare 1936 RCA set and a mass-produced model of the early 1950's. I also discovered that some clubs and current publications primarily de-

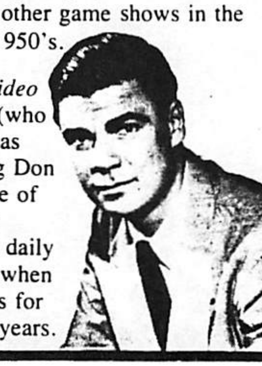
COLLECTORS' INFORMATION

A Flick of the Switch by Morgan E. McMahon, published by Vintage Radio (hard cover, $10.95; handbook, $8.95), contains a chapter on old TV and radio receivers with photographs, model numbers, and dates of a wide variety of sets.

A Flick of the Switch may be ordered from Vintage Radio, a mail order book service primarily for collectors of old radios and phonographs. Write to: Vintage Radio, Box 2054, Palo Verdes Peninsula, California 90274.

KTLA's *Space Patrol* is one of television's first space adventure series. Standing are Ken Mayer, Virginia Hewitt, and Nina Bara. Seated are Lynn Osborne and Ed Kemmer, who plays commander Buzz Corey. For eight years, these actors will do a fifteen-minute, live local show five days a week; a thirty-minute, live network show three times a week; and a live, weekly radio show.

Network daytime TV began in the late forties with Bert Parks hosting a game show called *What's My Name?* from Chicago. Parks will establish himself further by hosting the popular *Break the Bank* and other game shows in the 1940's and 1950's.

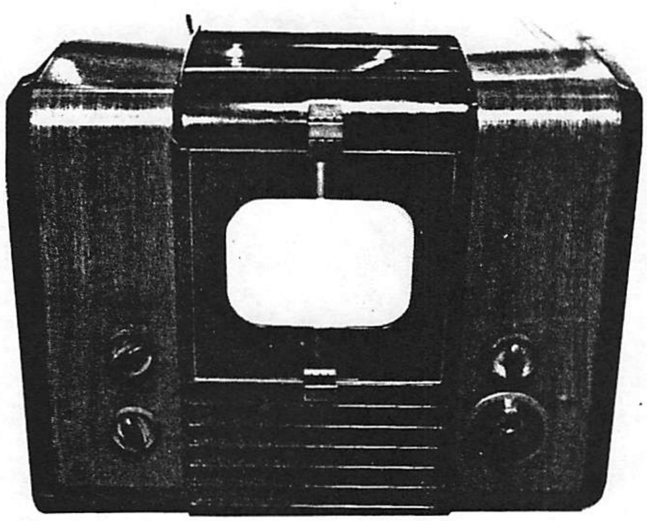


Captain Video and His Video Rangers, with Al Hodge (who replaced Richard Coogan as Captain Video) and young Don Hastings, ushers in a wave of TV sci-fi shows. Dumont broadcasts this serial on a daily basis from 1949 to 1953, when it becomes a weekly series for three more action-packed years.

voted to antique radio collecting are now beginning to show an interest in early television too. Many of the sets in my collection were obtained through these publications, as well as by hunting through swap-meets and at second-hand stores.

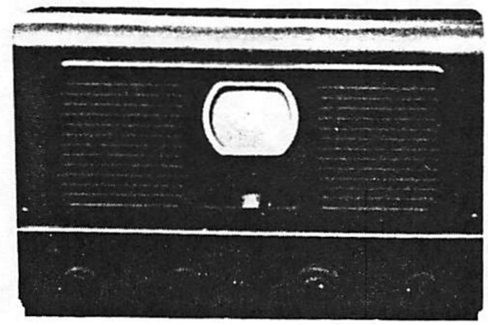
Although pre-1950 television sets can still be found at discount prices, it is very unlikely you will find them in working condition. Admirers of your newest acquisition will inevitably expect a demonstration, but unless you are electronically adept, repairs can be costly.

Extraordinary finds are not impossible, as a friend of mine proved recently by purchasing a 1936 RCA experimental receiver for forty dollars. My latest acquisition required exhausting legwork and a larger investment, but the result was a rewarding addition to my collection. It is an RCA model TRK-12 receiver that was used as a demonstration set for television's public debut at the 1939 New York World's Fair.



A 1946 television receiving set, the RCA 621 TS.

"Extraordinary finds are not impossible."



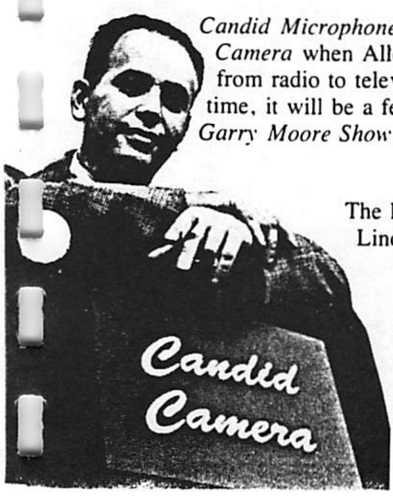
A 1948 Pilot-Candid TV receiver rescued from a junk pile.

These early RCA receivers are distinctive in appearance because the screen is in a horizontal position; it was viewed through a reflecting mirror positioned above, on the lid of the cabinet.

Aspiring collectors should note that these early television sets, manufactured before World War II, are not easy to find. I know of only a handful, most of which are in the hands of a very few devoted owners.

It is my hope that by increasing public awareness and interest in early television, many sets still hiding in garages and basements will surface and be saved from destruction. Perhaps you may be the next to locate a rare 1930's television set, or one of the elusive scanning-disk receivers of the 1920's, and thereby preserve a long neglected part of our American heritage.

Kent Warner is a TV costume designer who thinks museums should take an interest in television.



Candid Microphone becomes *Candid Camera* when Allen Funt switches from radio to television. For a short time, it will be a feature on *The Garry Moore Show*.

The husband and wife team of Peter Lind Hayes and Mary Healy enters TV via the 1949 variety show, *Inside U.S.A.* The following year, they will interview celebrities on *The Stork Club*. They will be visible on television well into the 1960's.

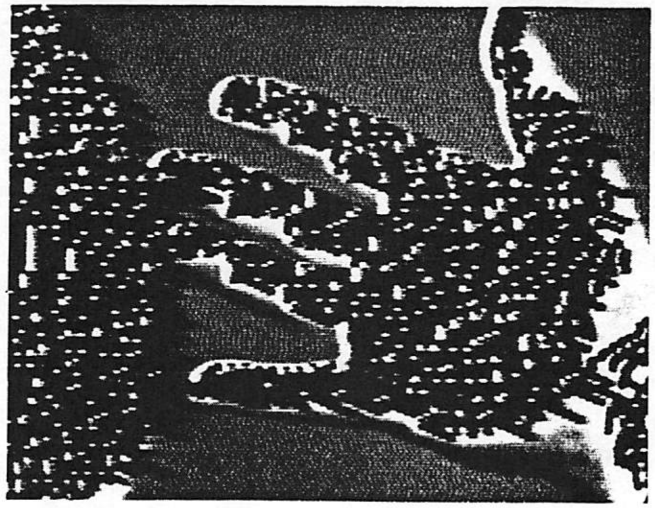
VIDEO ART

JOHN G. HANHARDT

The history of video art reads as a continual challenge to the traditional definition of the medium offered by broadcast television. Since its introduction in the late 1940's, commercial television has rapidly become a mass medium, plugged into virtually every household, and playing an active role in shaping the public's consciousness. The television set is a central piece of household furniture around which the family organizes its schedule and movements. Like a giant electronic night-light, the television set offers a kind of security, playing for hours on end, its screen emitting a glow that requires little concentration yet demands passive acceptance of its messages. Commercial television sells its air time to advertisers, and its programming provides a context and support for commercial messages. Even public television primarily presents traditional narratives and seldom takes risks except in specific issues of content and subject matter. The presentation of a visual abstraction, time of silence, or avant-garde narrative devices, is considered too high a risk.

The term *video art* describes a wide range of work produced in this electronic medium independent of the restrictions in form and content of commercial television. Video as a medium for personal creative expression was largely made possible by the introduction of the Sony Portapak video system in 1968. This system utilizes a handheld video camera and a portable, battery-powered, tape recorder. Unlike studio equipment which uses two-inch videotape, the portable unit employs one-half inch tape. The resulting lightweight and relatively inexpensive video system

"Video as an art form . . . exists within the context of the contemporary, modernist, and post-modernist art scene."



Video artists Woody and Steina Vasulka taped Vocabulary *in color in 1973–74. It uses representational and abstract images to create a mysterious electronic experience.*

has changed the previously held notion that television was the exclusive preserve of what we traditionally see on our home TV sets. Beginning in the 1940's, when 16-mm film and cameras were developed and mar-

The Arthur Murray Dance Party, featuring dance contests and instruction, with hostess Kathryn Murray and choreographer June Taylor, debuts on Dumont in 1950. From 1952 to 1957, it will serve as a summer replacement show, and will resume as a weekly show from 1958 to 1960 on NBC.

Saturday Night Revue changes its name to *Your Show of Shows*, a title it will keep until it goes off the air in 1954.

keted a similar situation occurred in filmmaking. Since then independent filmmaking has grown rich and varied; it is now a personal film art created outside the production, distribution, and exhibition network of the commercial Hollywood entertainment film.

Video as an art form essentially was created in the late 1960's, and exists within the context of the contemporary, modernist, and post-modernist art scene. This means that video shares many concepts with painting, sculpture, music, performance, theater, film, and the recent mutations, permutations, and breakdowns of these traditional categories. But the situation for the video artist is complicated by the fact that, unlike the other arts, video does not have a past to develop from or react against. Rather it has the definition of the medium imposed by the dominant "presence" of commercial television. Furthermore, while video art is strongly influenced by and is an integral part of the contemporary visual arts, it does not enjoy extensive economic and critical support, and is not a commodity in the art market, like a painting or work of sculpture. Except for some pieces produced by artists in conjunction with public television "experimental laboratories," and an occasional showing on public and cable television, video art is largely shown in an increasing number of museums, colleges and universities, galleries, independent exhibition spaces, and artists' lofts.

The range of work and quality of achievement of contemporary video is enormous. One area of experimentation has been the use of multiple monitors to create an environment of video images. Juan Downey's *Video Trans Americas,* for example, created a metaphor in video for the process of communication between cultures. Upon entering the exhibition space, the viewer saw monitors grouped at specific points representing the north-south, east-west axes, displaying images of cultures and places that alternated in a

"Video art challenges . . . the viewer by opening up new ways of perceiving . . . the world."

rhythmic pattern between the screens. In the center of the room a film image of a group of monitors was projected onto the floor from the ceiling. The viewer could walk onto the "screen" on the floor, and be seen live on one of the monitors placed in the room. Thus the viewer participated in the piece and actually became part of it as the video camera placed him in the video environment.

Woody and Steina Vasulka's video uses a single monitor to display a wide range of strong and mysterious abstract and representational images created out of the electronic process itself. Brian Connell's *La Lucha Final,* a single-channel video piece, is a treatment of the illusion in images we think we understand, namely commercial television's interpretation of news events. A political work employing modernist narrative devices, this videotape combines commentary with images to create a displacement of meaning, confounding fact and fiction. It ultimately asks questions about how we perceive the world through television's recording and interpretation of events.

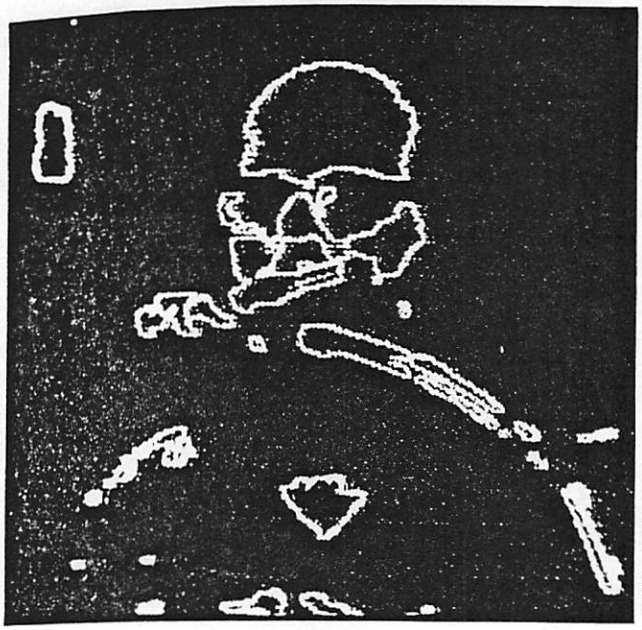
Unlike broadcast television, the best of video art demands that viewers concentrate and reflect upon what they have experienced. Video art challenges and ultimately satisfies the viewer by opening up new ways of perceiving and new perceptions of the world. It takes nothing for granted, least of all the medium of video and the way we relate to it.

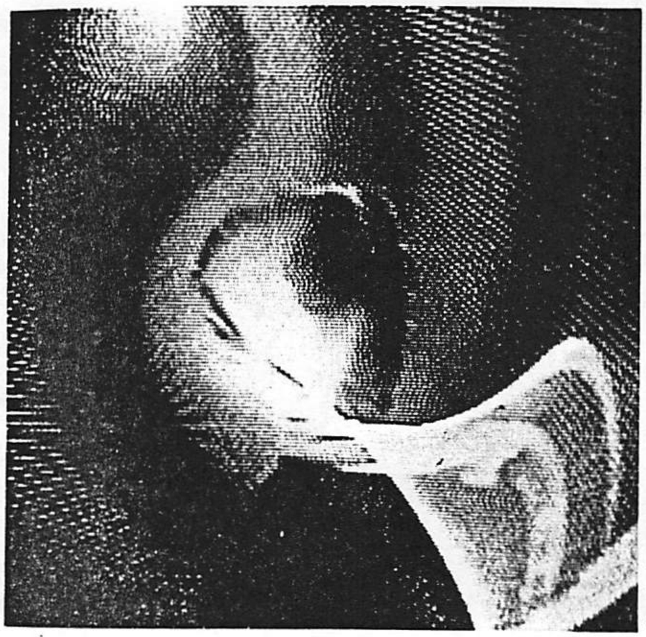
John G. Hanhardt is curator of Film and Video at the Whitney Museum of American Art in New York City.

CBS deals its competitors a crushing blow when it makes a deal with several big-name radio stars. Jack Benny arrives in 1950 and stays fourteen years. Here Benny is with his wife, Mary Livingston, a semi-regular on his program.

Benny's close friends, the great vaudeville team George Burns and Gracie Allen, bring their radio show to CBS where it will delight viewers for eight years. Here Burns tricks his announcer, Harry Von Zell.

Amos 'n' Andy, which started as an NBC radio show in 1929, is sold to CBS-TV in 1951. Freeman Gosden and Charles Correll, the white men who starred in and created the radio version, give way to Spencer Williams and Alvin Childress in the title roles. *Amos 'n' Andy* is the first TV show to have an all-black cast.


Davidson Gigliotti


Mary Lucier

Nam June Paik Comments:

"I have treated the TV screen as a canvas, and proved it can be a superior canvas. From now on, I will treat the cathode ray as a paper and pen. . . . If Joyce lived today, surely he would have written *Finnegan's Wake* on videotape, because of the vast possibility of manipulation in magnetic information storage."

"Plato thought the word, or the conceptual, expresses the deepest thing. St. Augustine thought the sound, or the audible, expresses the deepest thing. Spinoza thought the vision, or the visible, expresses the deepest thing. This argument is settled for good. TV commercials have all three."

"Tolstoy spent 20 pages for the description of Anna Karenina and Flaubert 30 pages for Madame Bovary. . . . What they needed was simply a Polaroid camera."

"TV cameras are following so busily the latest spots of violence that kids, who receive most of their education from TV, think that such noble countries as Switzerland and Norway are chunks of real estate lying somewhere in the Milky Way or at best beyond Madagascar. How can we teach about peace while blocking out one of the few existing examples from the screen?"

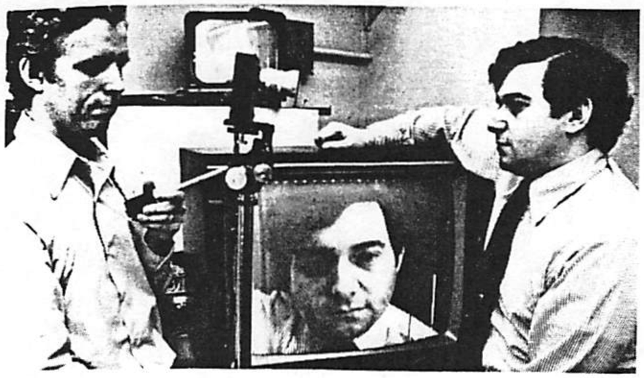
"My experimental TV is not always interesting but not always uninteresting. [It is] like nature, which is beautiful not because it changes beautifully but simply because it changes."

"Nobody had put two frequencies into one place, so (in 1962) I do just that, horizontal and vertical, and this absolutely new thing comes out. I make mistake after mistake, and it comes out positive. That is the story of my whole life."

Strike It Rich provides luckless contestants with a forum to explain why they could use donations from home viewers. The contestant with the most pitiful tale of the day is also awarded the "heartline," a large cash prize, by host Warren Hull (R). Sailor B. M. Frazer seems too happy for this show.

Jacqueline Susann hosts her own thirty-minute talk show for Dumont in 1951. In 1953 she will do the same for ABC.

March 5, 1951. *Mr. Wizard* debuts, and at last there is an educational program that is truly popular with the junior set. For the next fourteen years, Don Herbert will prove to boys and girls that science can be fun as well as interesting.



RCA experimenters with a small, solid state camera.

GLOSSARY

Coaxial cable. A special kind of cable consisting of a small inner conductor separated by insulation from a concentric outer conductor. It is capable of carrying the wide band-width required to transmit television signals over long distances.

Film camera. A device used to convert the optical images on a motion picture film (or on slides) to a television signal.

Hookup. A system of coaxial cable, microwave, or satellite connections between two or more stations that allows them simultaneously to telecast a program originating at one point.

Iconoscope. The very first type of pickup tube. Invented by V. K. Zworykin in 1923, used in most prewar television systems.

Image orthicon. Later type of pickup tube. Used in most systems from 1947 to the mid-1960's.

Kinescope. Technical term for the picture tube in the home receiver (or monitor). In this tube an electron scanning beam causes the color dots on the kinescope face to fluoresce in accordance with the received signal and thus to reproduce the scene at the transmitting point.

Live camera. A device used to pick up the optical image of a live event (sports, news, studio program) and convert it to a television signal. It includes a lens system, a pickup tube (or tubes), an amplifier, and control circuits.

Microwave system. A chain of relay stations, located about twenty-five miles apart, which are used to send television signals over long distances (e.g., coast-to-coast).

Pickup Tube. An electron tube in the camera which electronically scans the image and generates an electrical signal proportioned to the light intensity at each spot in the picture.

Plumbicon®. An improved pickup tube introduced in 1962 and used in most broadcast TV systems today.

Receiving antenna. An arrangement of rodlike elements (usually mounted on the house top) that picks up signal radiated from the transmitting antenna and feeds it to the receiver.

Satellite. A relay station in the sky located in geosynchronous orbit, it receives signals from one point on earth and relays (retransmits) them to another point a continent, or an ocean, away.

STL (Studio Transmitter Link). A special telephone line, coaxial cable, or microwave system used to send the television signal from the studio to the transmitter site.

Studio switcher. A large console with monitors, indicator lights, and rows of push-button switches. Used for observing pictures from various sources (remote points, various studios, VTR's, film cameras) and selecting one to be sent to the transmitter for telecast.

Transmitter. A large piece of equipment that amplifies the television signal received from the studio to high power, converts it to VHF (very high frequency) or UHF (ultra high frequency) and feeds it to the antenna.

During the war, Dumont is the only network to continue extensive commercial programming. Paul Winchell and Jerry Mahoney appear on WABD, New York.

1946. The war over, television begins to dominate domestic life. Within the next five years the total number of home TV sets in America will rise from ten thousand to twelve million. In the 1950's, forty million Americans will be born and seventy million sets built.

Heavyweight boxer Ezzard Charles poses with Gorgeous George, the man chiefly responsible for "modernizing" wrestling. Spectator sports that demand sophisticated camera work must take a back seat to boxing and wrestling on early TV.

THE DREAMERS WHO MADE TELEVISION

Modern television is the creation of many men. But two names stand out above all others—Vladimir Zworykin and David Sarnoff. Both were born in Russia, came to this country as youths, became U.S. citizens, and flourished under the American system. In 1923 Zworykin invented the iconoscope, the first camera tube, and in 1928 the kinescope, the first picture tube. In 1929 he described to David Sarnoff his idea for using these as the basis of an all-electronic television system. Sarnoff, then vice-president and general manager of RCA, was a man of great prescience. He quickly grasped the possibilities of electronic television and agreed to provide the money and manpower to develop it.

The vision that Zworykin and Sarnoff shared came true. But not until numerous obstacles had been surmounted. It took many years and millions of dollars. Sarnoff was acutely conscious of the protracted struggle, and often spoke of it. One such occasion was at the 1944 convention of the Television Broadcasters Association, when both he and Dr. Zworykin were given awards. Zworykin, who received his first, was hailed as the man who made television possible. In accepting, Zworykin said, "I was just the dreamer. I went to David Sarnoff with my idea and he agreed to back me. It was he who made television possible." When it became Sarnoff's turn he referred to Zworykin's statement and said, "as I remember it, Dr. Zworykin came to me with his idea for an electronic system. I asked him how much it would cost and how long it would take. He said it would cost about a hundred thousand dollars and might take a year and a half. That was fifteen years, and ten million dollars, ago—and I believed him! Who was the dreamer?"

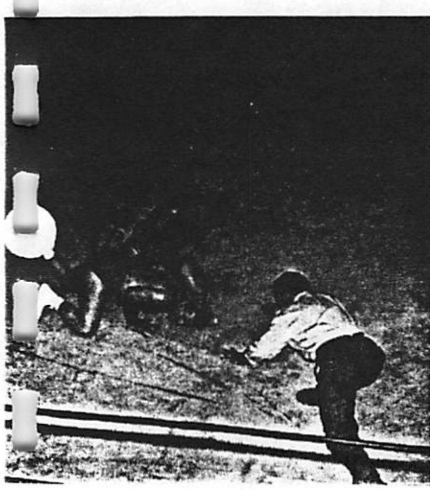
As things turned out they were both still dreaming. It was to be another five years, and cost RCA some $40 million more, before black-and-white television became profitable in 1949. And it took an additional sixteen years (until 1965), and another $100 million of RCA's money, before color television was fully established.

Transmitting antenna. A turnstilelike arrangement of pipes and rods (usually mounted on a tall tower) that radiates the high-power television signal in all directions.

TV receiver. The device in the home that amplifies the received signals, decodes them, and displays the program on the picture tube that it houses.

Vidicon. A less expensive type pickup tube used in many industrial and educational type TV cameras.

VTR (Video Tape Recorder). A device used to record a television signal on magnetic tape so that it can be telecast at a later time or sent to other stations, as in syndication.



By the end of the decade, wrestling will become so popular that it runs every night in some cities. *TV Guide* will set up a department to compile match results. Dumont announcer Dennis James and wrestlers such as Gorgeous George, Baron Leone, the Zebra Kid, Ricki Starr, and Antonino Rocca become household words.

Inexpensively produced game shows, often with children as panelists, fill television schedules in the late 1940's.

11. GLOSSARY

AES/EBU – Informal name for a digital audio standard established jointly by the Audio Engineering Society and European Broadcasting Union organizations.

algorithm – A set of rules or processes for solving a problem in a finite number of steps.

aliasing – Defects in the picture typically caused by insufficient sampling or poor filtering of digital video. Defects are typically seen as jaggies on diagonal lines and twinkling or brightening in picture detail.

analog – An adjective describing any signal that varies continuously as opposed to a digital signal that contains discrete levels representing the binary digits 0 and 1.

asynchronous – A transmission procedure that is not synchronized by a clock.

A-to-D Converter (analog-to-digital) – A circuit that uses digital sampling to convert an analog signal into a digital representation of that signal.

bandwidth – 1. The difference between the upper and lower limits of a frequency, often measured in megahertz (MHz). 2. The complete range of frequencies over which a circuit or electronic system can function with less than a 3 dB signal loss. 3. The information carrying capability of a particular television channel.

baseline shift – A form of low-frequency distortion resulting in a shift in the DC level of the signal.

bit – A binary representation of 1 or 0. One of the quantized levels of a pixel.

bit parallel – Byte-wise transmission of digital video down a multi-conductor cable where each pair of wires carries a single bit. This standard is covered under SMPTE125M, EBU 3267-E and CCIR 656.

bit serial – Bit-wise transmission of digital video down a single conductor such as coaxial cable. May also be sent through fiber optics. This standard is covered under CCIR 656.

bit slippage – 1. Occurs when word framing is lost in a serial signal so that the relative value of a bit is incorrect. This is generally reset at the next serial signal, TRS-ID for composite and EAV/SAV for component. 2. The erroneous reading of a serial bit stream when the recovered clock phase drifts enough to miss a bit. 3. A phenomenon which occurs in parallel digital data buses when one or more bits gets out of time in relation to the rest. The result is erroneous data. Differing cable lengths is the most common cause.

bit stream – A continuous series of bits transmitted on a line.

blocking – Occurs in a multistage routing system when a destination requests a source and finds that source unavailable. In a tie line system, this means that a destination requests a tie line and receives a "tie line busy" message, indicating that all tie lines are in use.

BNC – Abbreviation of "baby N connector." A cable connector used extensively in television.

byte – A complete set of quantized levels containing all of the bits. Bytes consisting of 8 to 10 bits per sample are typical.

cable equalization – The process of altering the frequency response of a video amplifier to compensate for high-frequency losses in coaxial cable.

CCIR – International Radio Consultative Committee, an international standards committee.

CCIR-601 – An international standard for component digital television from which was derived SMPTE 125M (was RP-125) and EBU 3246E standards. CCIR defines the sampling systems, matrix values, and filter characteristics for both Y, B-Y, R-Y and RGB component digital television.

CCIR-656 – The physical parallel and serial interconnect scheme for CCIR-601. CCIR 656 defines the parallel connector pinouts as well as the blanking, sync, and multiplexing schemes used in both parallel and serial interfaces. Reflects definitions in EBU Tech 3267 (for 625 line signals) and in SMPTE 125M (parallel 525) and SMPTE 259M (serial 525).

channel coding – Describes the way in which the 1s and 0s of the data stream are represented on the transmission path.

character generator (CG) – A computer used to generate text and sometimes graphics for video titles or captions.

clock jitter – Timing uncertainty of the data cell edges in a digital signal.

clock recovery – The reconstruction of timing information from digital data.

coaxial cable – A transmission line with a concentric pair of signal carrying conductors. There is an inner conductor and an outer conductor metallic sheath. The sheath aids in preventing external radiation from affecting the signal on the inner conductor and minimizes signal radiation from the transmission line.

coding – Representing each level of a video signal as a number, usually in binary form.

coefficients – A number (often a constant) that expresses some property of a physical system in a quantitative way.

component analog – The unencoded output of a camera, videotape recorder, etc., consisting of three primary color signals: red, green, and blue (RGB) that together convey all necessary picture information. In some component video formats, these three components have been translated into a luminance signal and two color difference signals, for example, Y, B-Y, R-Y.

component digital – A digital representation of a component analog signal set, most often Y, B-Y, R-Y. The encoding parameters are specified by CCIR 601. The parallel interface is specified by CCIR 656 and SMPTE 125M (1991).

composite analog – An encoded video signal, such as NTSC or PAL video, that includes horizontal and vertical synchronizing information.

compression artifacts – Compacting of a digital signal, particularly when a high compression ratio is used, may result in small errors in the decompressed signal. These errors are known as "artifacts," or unwanted defects. The artifacts may resemble noise (or edge "busyness") or may cause parts of the picture, particularly fast moving portions, to be displayed with the movement distorted or missing.

composite digital – A digitally encoded video signal, such as NTSC or PAL video, that includes horizontal and vertical synchronizing information.

contouring – Video picture defect due to quantizing at too coarse a level.

D1 – A component digital video recording format that uses data conforming to the CCIR-601 standard. Records on 19mm magnetic tape. (Often used incorrectly to indicate component digital video.)

D2 – A composite digital video recording format that uses data conforming to SMPTE 244M. Records on 19mm magnetic tape. (Often used incorrectly to indicate composite digital video.)

D3 – A composite digital video recording format that uses data conforming to SMPTE 244M. Records on 1/2" magnetic tape.

delay – The time required for a signal to pass through a device or conductor.

demultiplexer (demux) – A device used to separate two or more signals that were previously combined by a compatible multiplexer and transmitted over a single channel.

deserializer – A device that converts serial digital information to parallel.

differential gain – A change in chrominance amplitude of a video signal caused by a change in luminance level of the signal.

differential phase – A change in chrominance phase of a video signal caused by a change in luminance level of the signal.

digital word – The number of bits treated as a single entity by the system.

discrete – Having an individual identity. An individual circuit component.

dither – Typically a random, low-level signal (oscillation) which may be added to an analog signal prior to sampling. Often consists of white noise of one quantizing level peak-to-peak amplitude.

dither component encoding – A slight expansion of the analog signal levels so that the signal comes in contact with more quantizing levels. The results are smoother transitions. This is done by adding white noise (which is at the amplitude of one quantizing level) to the analog signal prior to sampling.

drift – Gradual shift or change in the output over a period of time due to change or aging of circuit components. Change is often caused by thermal instability of components.

D-to-A converter (digital to analog) – A device that converts digital signals to analog signals.

DVTR – Abbreviation of *digital videotape recorder.*

EAV – End of active video in component digital systems

EBU – European Broadcasting Union. An organization of European broadcasters that, among other activities, produces technical statements and recommendations for the 625/50 line television system.

EBU TECH.3267-E – The EBU recommendation for the parallel interface of 625 line digital video signal. A revision of the earlier EBU Tech.3246-E, which in turn was derived from CCIR-601 and contributed to CCIR-656 standards.

EDH (error detection and handling) – Proposed SMPTE RP-165 for recognizing inaccuracies in the serial digital signal. It may be incorporated into serial digital equipment and employ a simple LED error indicator.

Equalization (EQ) – Process of altering the frequency response of a video amplifier to compensate for high-frequency losses in coaxial cable.

embedded audio – Digital audio is multiplexed onto a serial digital data stream.

encoder – In video, a device that forms a single, composite color signal from a set of component signals.

error concealment – A technique used when error correction fails (see error correction). Erroneous data is replaced by data synthesized from surrounding pixels.

error correction – A scheme that adds overhead to the data to permit a certain level of errors to be detected and corrected.

eye pattern – A waveform used to evaluate channel performance.

field-time (linear) distortion – An unwarranted change in video signal amplitude that occurs in a time frame of 16 ms.

format conversion – The process of both encoding/decoding and resampling of digital rates.

frequency modulation – Modulation of sine wave or "carrier" by varying its frequency in accordance with amplitude variations of the modulating signal.

frequency response rolloff – A distortion in a transmission system where the higher frequency components are not conveyed at their original full amplitude and possible loss of color saturation.

gain – Any increase or decrease in strength of an electrical signal. Gain is measured in terms of decibels or number of times of magnification.

group delay – A signal defect caused by different frequencies having differing propagation delays (delay at 1 MHz is different from delay at 5 MHz).

horizontal interval (horizontal blanking interval) – The time period between lines of active video

interpolation – In digital video, the creation of new pixels in the image by some method of mathematically manipulating the values of neighboring pixels.

I/O – Abbreviation of *input/output.* Typically refers to sending information or data signals to and from devices.

jaggies – Slang for the stair-step aliasing that appears on diagonal lines. Caused by insufficient filtering, violation of the Nyquist Theory, and/or poor interpolation.

jitter – An undesirable random signal variation with respect to time.

MAC – Multiplexed Analog Component video. This is a means of time multiplexing component analog video down a single transmission channel such as coax, fiber or a satellite channel. Usually involves digital processes to achieve the time compression.

microsecond (μ) – One millionth of a second: 1×10^{-6} or 0.000001 second.

Miller squared coding – A DC-free channel coding scheme used in D-2 VTRs.

MPEG-2 – Motion pictures expert group. An international group of industry experts set up to standardize compressed moving pictures and audio.

multi-layer effects – A generic term for a mix/effects system that allows multiple video images to be combined into a composite image.

multiplexer (mux) – Device for combining two or more electrical signals into a single, composite signal.

nanosecond (ns) – One billionth of a second: 1×10^{-9} or 0.000000001 second.

NICAM (near instantaneous companded audio multiplex) – A digital audio coding system originally developed by the BBC for point-to-point links. A later development, NICAM 728 is used in several European countries to provide stereo digital audio to home television receivers.

nonlinear encoding – Relatively more levels of quantization are assigned to small amplitude signals, relatively fewer to the large signal peaks.

nonlinearity – Having gain vary as a function of signal amplitude.

NRZ – Non return to zero. A coding scheme that is polarity sensitive. 0 = logic low; 1 = logic high.

NRZI – Non return to zero inverse. A video data scrambling scheme that is polarity insensitive. 0 = no change in logic; 1 = a transition from one logic level to the other.

NTSC (National Television Systems Committee) – Organization that formulated standards for the NTSC television system. Now describes the American system of color telecasting which is used mainly in North America, Japan, and parts of South America.

Nyquist sampling theorem – Intervals between successive samples must be equal to or less than one-half the period of highest frequency.

orthogonal sampling – Sampling of a line of repetitive video signal in such a way that samples in each line are in the same horizontal position.

PAL (Phase Alternate Line) – The name of the color television system in which the V component of burst is inverted in phase from one line to the next in order to minimize hue errors that may occur in color transmission.

parallel cable – A multi-conductor cable carrying simultaneous transmission of data bits. Analogous to the rows of a marching band passing a review point.

patch panel – A manual method of routing signals using a panel of receptacles for sources and destinations and wire jumpers to interconnect them.

peak to peak – The amplitude (voltage) difference between the most positive and the most negative excursions (peaks) of an electrical signal.

phase distortion – A picture defect caused by unequal delay (phase shifting) of different frequency components within the signal as they pass through different impedance elements – filters, amplifiers, ionospheric variations, etc. The defect in the picture is "fringing"—like diffraction rings—at edges where the contrast changes abruptly.

phase error – A picture defect caused by the incorrect relative timing of a signal in relation to another signal.

phase shift – The movement in relative timing of a signal in relation to another signal.

pixel – The smallest distinguishable and resolvable area in a video image. A single point on the screen. In digital video, a single sample of the picture. Derived from the words *picture element.*

PRBS – Pseudo random binary sequence.

production switcher (vision mixer) – A device that allows transitions between different video pictures. Also allows keying and matting (compositing).

propagation delay (path length) – The time it takes for a signal to travel through a circuit, piece of equipment, or a length of cable.

quantization – The process of converting a continuous analog input into a set of discrete output levels.

quantizing noise – The noise (deviation of a signal from its original or correct value) which results from the quantization process. In serial digital, a granular type of noise only present in the presence of a signal.

rate conversion – 1. Technically, the process of converting from one sample rate to another. The digital sample rate for the component format is 13.5 MHz; for the composite format it is either 14.3 MHz for NTSC or 17.7 MHz for PAL. 2. Often used incorrectly to indicate both resampling of digital rates and encoding/decoding.

reclocking – The process of clocking the data with a regenerated clock.

resolution – The number of bits (four, eight, ten, etc.) determines the resolution of the digital signal.

4-bits = a resolution of 1 in 16

8-bits = a resolution of 1 in 256

10-bits = a resolution of 1 in 1024

Eight bits is the minimum acceptable for broadcast TV.

RP-125 – See SMPTE 125M

routing switcher – An electronic device that routes a user-supplied signal (audio, video, etc.) from any input to any user-selected output(s).

sampling – Process where analog signals are measured, often millions of times per second for video.

sampling frequency – The number of discrete sample measurements made in a given period of time. Often expressed in megahertz for video.

SAV – Start of active video in component digital systems

scope – Short for oscilloscope (waveform monitor) or vectorscope, devices used to measure the television signal.

scrambling – 1. To transpose or invert digital data according to a prearranged scheme in order to break up the low-frequency patterns associated with serial digital signals. 2. The digital signal is shuffled to produce a better spectral distribution.

serial digital – Digital information that is transmitted in serial form. Often used informally to refer to serial digital television signals.

serializer – A device that converts parallel digital information to serial digital.

SMPTE (Society of Motion Picture and Television Engineers) – A professional organization that recommends standards for the television and film industries.

SMPTE 125M (was RP-125) – The SMPTE recommended practice for bit parallel digital interface for component video signals. SMPTE 125M defines the parameters required to generate and distribute component video signals on a parallel interface.

SMPTE 244M – The SMPTE recommended practice for bit parallel digital interface for composite video signals. SMPTE 244M defines the parameters required to generate and distribute composite video signals on a parallel interface.

SMPTE 259M – The SMPTE recommended practice for 525 line serial digital component and composite interfaces.

still store – Device for storage of specific frames of video.

synchronous – A transmission procedure by which the bit and character stream are slaved to accurately synchronized clocks, both at the receiving and sending end.

sync word – A synchronizing bit pattern, differentiated from the normal data bit patterns, used to identify reference points in the television signal; also to facilitate word framing in a serial receiver.

telecine – A device for capturing movie film as a video signal.

temporal aliasing – A visual defect that occurs when the image being sampled moves too fast for the sampling rate. A common example is wagon wheels that appear to rotate backwards.

time base corrector – Device used to correct for time base errors and stabilize the timing of the video output from a tape machine.

TDM (time division multiplex) - The management of multiple signals on one channel by alternately sending portions of each signal and assigning each portion to particular blocks of time.

time-multiplex – In the case of CCIR-601, a technique for transmitting three signals at the same time on a group of parallel wires (parallel cable).

TRS – Timing reference signals in composite digital systems (four words long)

TRS-ID (timing reference signal identification) – A reference signal used to maintain timing in composite digital systems. It is four words long.

truncation – Deletion of lower significant bits on a digital system. Usually results in digital noise.

VTR (video tape recorder) – A device which permits audio and video signals to be recorded on magnetic tape.

waveform – The shape of an electromagnetic wave. A graphical representation of the relationship between voltage or current and time.

word – See byte

125M – See SMPTE 125M

4:2:2 – A commonly-used term for a component digital video format. The details of the format are specified in the CCIR-601 standard document. The numerals 4:2:2 denote the ratio of the sampling frequencies of the single luminance channel to the two color difference channels. For every four luminance samples, there are two samples of each color difference channel. See *CCIR-601*.

4fsc – Four times subcarrier sampling rate used in composite digital systems. In NTSC this is 14.3 MHz. In PAL this is 17.7 MHz.

AUDIO TERMINOLOGY

1. **ACOUSTICS** Acoustics is the study of sound and its interaction with the human hearing mechanism.

2. **AMBIENCE** Ambience refers to the acoustical qualities of a listening space. Reverberation, echoes, background noise, etc., are components of ambience.

3. **AMPLIFIER** A device for increasing the amplitude of the voltage, current, impedence or power of a signal. The amount of amplification that an amplifier provides is called its GAIN. The gain is the ratio of its input signal level to its output signal level.

4. **ANALOG** An audio signal is an electrical replica, or analog, of the WAVEFORM of the sound it represents. The voltage of the signal varies up and down (negatively & positively, in electrical terminlogy) the same way as the SOUND PRESSURE varies up and down at the microphone.

4. **ANALOG-TO-DIGITAL CONVERTER, abbr. ADC or A/D** In DIGITAL audio systems, the analog audio signal must first be converted to digital form before it can be further processed. This entails SAMPLING the signal at very short successive time intervals, and converting the height of each sample to a digital word, which is simply a BINARY number indicating the AMPLITUDE of the WAVEFORM at that instant.

5. **BANDWIDTH** Bandwidth is literally a frequency span. For instance, the human voice can be transmitted with good intelligibility if the FREQUENCY RESPONCE of the transmission chain extends from about 100 Hz to about 3,000 Hz. Thus, a 2,900 Hz bandwidth is needed to transmit voice. This is about what the standard telephone system attains. The full audio bandwidth of human hearing however, is generally considered to be about 20 KILOHERTZ (20 KHz).

6. **BASS** Bass is that portion of the audible FREQUENCY which encompasses the lower PITCHES. The bass range is generally considered to be from 30 Hz or so up to 200 Hz.

AUDIO TERMINOLOGY

7. **BINARY** A Binary number system uses only two digits, as oppossed to the decimal system which uses ten. Binary numbers consist of a series of ones and zeros, and are easy to implement in computer systems because the presence of a voltage can indicate a 1, while the absence indicates a 0.

8. **BINARY DIGITS abbr. "BITS"**, are the numbers used in the binary number system. Bits are commonaly divided into groups of eight, which are called binary word, or BYTES.

9. **BIT RATE** The time rate at which bits are transmitted in a digital system, expressed in bits per second (bps). The bit rate of the compact disk is 4.3218 million bps.

10. **BYTE** In digital systems, the binary BITS are grouped into words. A BYTE is a binary word of eight bits. The information-storage capability of digital systems is measured in bytes or kilobytes.

11. **CD-ROM** Acronym for compact disc read only memory. The compact disc medium modified to store text and picture information rather than audio. This is analogous to a magnetic disc in a computer, except the storage is by optical means.

12. **DAT** A digital audio tape (DAT) is a cassette-like format for recording audio digitally on magnetic tape.

13. **DECIBEL or dB** Literally one-tenth of a bel, named after Alexander Graham Bell. The bel is defined as the common logerithm of the ratio of two powers. Thus, two powers, one of which is ten times the other, will differ by one bel; 10 WATTS are 1 bel higher in level than 1 WATT. A 360-horsepower car is 1 bel more powerful than a 36 horsepower motorcycle. Any power ratio can be expressed in bels.

AUDIO TERMINOLOGY

Because the bel is a ratio of 10 and this is a rather large ratio, it is convenient to divide in into tenths of bels, or decibels. Thus the decibel is ten times the common log of the ratio of two powers:

dB = 10 log P/P where P stands for the two powers being compared.

Another way to think of decibels is to think in terms of percentages. We all know what 10 percent means, and nobody thinks of percentages as being quantities of anything. A decibel is nothing more than a power change of 27%, 3 dB is a power change of 100% etc.

Because decibels refer to a ratio, it can describe many different types of audio relationships. For our purposes, we will refer to decibels as a measure of the amplitude of a waveform or the loudness of a sound, unless otherwise noted.

14. **DIGITAL-TO ANALOG CONVERTER, OR DAC** The component within a digital audio device which converts binary digital words into an ANALOG signal that can be amplified and sent to a loudspeaker, etc. The DAC is the lst link in the digital chain.

15. **DYNAMIC RANGE** The dynamic range of a sound is the ratio of the strongest, or loudest, part to the weakest, or softest, part. It is measured in decibels (dB). A full orchestra may have a dynamic range of 90 dB, meaning the softest passages are 90 dB less powerful than the loudest ones. Dynamic range is a power ratio and has nothing to do with the absolute level of the sound.

Rarely is the dynamic range of an audio system as large as the dynamic range of an orchestra because of several factors. The inherant noise of the recording medium determines the softest possible recorded sound, and the maximum signal capacity of the system (clipping level) limits the the loudest possible sound.

16. **ECHO** Commonly used incorrectly to mean REVERBERATION, echo technically is a discrete sound reflection arriving at least 50 milliseconds after the direct sound.

AUDIO TERMINOLOGY

17. **EQUALIZER, EQUALIZATION** An equalizer, contrary to what its name implies, alters or distorts the relative strength of certain frequency ranges of an audio signal. It can be used to either boost or cut the desired frequency. Consumer type equalizers consist of the bass and treble tone controls.

18. **FLAT FREQUENCY RESPONSE** An amplifier, loudspeaker, microphone, etc., is said to have flat frequency response, or "flat response," if its output is at the same level for all frequencies of interest, provided its input also has uniform amplitude over the same frequency range. A flat system has the same gain or sensitivity at all frequencies of interest.

19. **FLETCHER-MUNSON EFFECT** In the early 1930's, Fletcher and Munson undertook to measure the sensitivity of the human hearing mechanism at different frequencies. At very low levels, the human ear is most sensitive to frequencies between 3 KHZ to 4 KHz, then the sensitivity drops off rapidly for lower frequencies and somewhat more slowly for higher frequencies. In other words, very soft sounds must be more powerful at frequencies lower and higher than 3 to 4 KHz to be heard. At louder levels however, the sensitivity curve flattenes out, meaning the human ear has much more uniform sensitivity at high levels.

20. **FOLEY EFFECTS** Sound effects in a motion picture or video that are produced by various mechanical devices operated by hand. Examples are slamming doors and creaking floors.

21. **FREQUENCY SEE HAND-OUT**

22. **FREQUENCY RESPONSE** Frequency response is a shortened way of stating the AMPLITUDE (gain, measured in dB) response verses frequency characteristic. It is usually presented as a graph or plot of the output of a device on the vertical axis versus the frequency on the horizontal axis. It is important to realize that frequency response is defined to be a characteristic of a system or a device, not a characteristic of a signal.

AUDIO TERMINOLOGY

23. **FUNDEMENTAL** The lowest FREQUENCY component in a complex periodic WAVEFORM. Any sound waveform which is perceived as having a musical PITCH is periodic, ie., it has a shape that repeats itself. This sound will have a series of HARMONICS, or OVERTONES which give the sound its defining character. An oboe has different harmonics than a flute playing the same note. The first HARMONIC is called the FUNDEMENTAL. The musical pitch of a sound is determined by the fundemental frequency.

24. **GAIN** The amount of increase in the power of a signal by an amplifier is called the power gain. It is simply the ratio of the output power to the input power and is expressed in decibels. Often used to refer to volume, loudness or level.

25. **HEADROOM** Headroom in an audio device is the difference in level present in a given signal and the maximum level the device can handle without noticeable distortion. Music exhibits very wide variations in level (softest to loudest) and any audio device must handle the maximum expected level in order to sound distortion free. In most audio equipment of high quality, the maximum signal level will be at least 10 dB above the maximum level shown on the meter, ie., there will be 10 dB of headroom.

26. **HERTZ, Hz** The internationally agreed upon symbol to indicate frequency in cycles per cycle. It is named after Heinrich Hertz, the famous German physicist who first investigated radio waves.

27. **LEVEL** The term level is loosely used when the magnitude of a signal is meant, usually refering to loudness or volume.

28. **LOUNDNESS** Loudness is a subjective attribute of a sound and cannot be quantified, except in a statistical sense. The loudness of a sound, especially a complex sound containing many frequencies, has no simple relation to its sound pressure level, and it is hopeless to try to measure relative loudness because our hearing is so complicated.

AUDIO TERMINOLOGY

29. **MID-RANGE** The frequency span of the middle of the audio range is usually considered to be from about 200 Hz to 2,000 Hz or so. Most of any music signal is in the mid-range as is human speech.

30. **MONO, OR MONAURAL** Literally, "one hearing". Mono refers to a sound system with only one channel, regardless of the number of speakers.

31. **NOISE** Noise can be defined as any unwanted sound which is not related to the wanted sound (if it is related, its called DISTORTION). The most common type of noise is called "random noise" because it is unpredictable from moment to moment. True random noise sounds like hissing and has no detectable pitch. An example of random noise is the sound heard in FM receivers when tuned off station.

Unfortunately, just about anything you do to a signal adds some noise to it. In copying a tape recording, the copy will have at least 3 dB more noise than the original, no matter how good the tape recorder.

Ambrose Bierce defines noise as "a stench to the ear. Undomesticated music. The chief product and authenticating sign of civilization."

32. **OCTAVE** An octave is a frequency ratio of 2:1. There is one octave between 100 Hz and 200 Hz, and also between 1,000 Hz and 2,000 Hz. Octaves are perceived as equal pitch intervals. To our ears, two frequencies an octave apart sound like the same note.

33. **OVERTONES** Overtones are tones produced by a musical instrument which are higher in FREQUENCY than the FUNDEMENTAL. All musical instruments produce comples sound WAVEFORMS which repeat at their fundemental frequency. The overtones determine the characteristic of the sound.

AUDIO TERMINOLOGY

34. **PINK NOISE** Pink Noise is a type of random noise which has a constant amount of energy in each octave band, as opposed to WHITE NOISE, which has equal energy per hertz. Because our hearing diminishes in the higher and lower frequency ranges, pink noise boosts those frequencies according to the Fletcher Munson curve so that the apparent sound of the pink noise is uniform to the human ear.

35. **PRESENCE** A boost in the frequency range between about 1 and 3 Kilohertz that makes music seem as though it is more "present" in the room.

36. **PURE TONE** A sound whose waveform is a SINE WAVE; or a SIGNAL with a single frequency and no harmonics.

37. **REFERENCE LEVEL** The reference level in an audio device is a SIGNAL level near the maximum possible for the device but low enough to ensure low distortion. To visually monitor the signal level in professional recording devices, a meter is used. This reference level was standardized in 1954 by the NAB and is that level at which a tone of 700 Hz will produce third HARMONIC DISTORTION of 1 percent. Professionally, a 1 KHz tone is used as the reference frequency and is set to read "0" on the vu meter.

38. **RESONANCE** A resonance is the tendency of a mechanical or electrical system to vibrate at a certain FREQUENCY when excited by an external force, and to keep vibrating after the excitation is removed. A bell is a good example.

38. **REVERBERATION** The remainder of sound that exists in a room after the source of the sound is stopped is called reverberation, sometimes mistakenly called "echo." The time of reverberation is defined as the time it takes for the SOUND PRESSURE LEVEL to decay by 60 dB.

AUDIO TERMINOLOGY

All rooms have some reverberation and it is the reverberation time that defines the room in acoustical terms, ie., a cathedral has much more reverb than does a small room for instance, and it is because of the reverb that we can tell the two spaces apart. The sound heard by a listener is a mixture of the direct sound from the source and reverberant sound from the space. Reverberant sound is diffuse, coming from random directions.

39. **SAMPLING** In a DIGITAL audio system, the audio signal must be fed into an ANALOG TO DIGITAL CONVERTER to be changed into a series of numbers for further processing. The first step is the sampling, where the instantaneous signal amplitude is determined at very short intervals of time. The sampling rate, which is the number of samples per second, must be uniform and precisely controlled. The professional sampling rate is 48 KHz per second. The consumer sampling rate is 44.1 kKHz per second which is the frequency used by compact disc recorders.

40. **SIGNAL** A signal is an electrical phenomenon, usually a voltage but sometimes a current, which contains desired information, as opposed to noise which is undesired. Audio signals are generally electrical analogs of the corresponding sound WAVEFORMS.

41. **SIGNAL-TO-NOISE RATIO** Signal to noise ratio is the ratio of the signal power at a certain reference point in a circuit to the noice power which would exist if the signal were removed. This ratio is expressed in decibels.

42. **SINE WAVE** The sine wave is the simplest possible periodic WAVEFORM. It consists of a single FREQUENCY and has a musical PITCH but a neutral TIMBRE or tone quality. It is called a sine wave because it has the same shape as the mathematical sine function. Sine waves are commonly used as test signals for audio equipment because they consist of only one frequency.

AUDIO TERMINOLOGY

43. **SOUND PRESSURE, SOUND PRESSURE LEVEL** A sound wave progressing through air causes the instantaneous air pressure at any given point to vary above and below the barometric pressure in accordance with the WAVEFORM of the sound. This variation in pressure is used as a quantitative measure of the strength of the sound, and is called sound pressure. The reference pressure is 0 dB on the scale and corresponds to the threshold of hearing at 1000 Hz for the normal human ear.

44. **STEREOPHONIC OR STEREO** In common usage, "stereo" has come to mean any sound system with two speakers. However, stereophonic refers to a system which provides the illusion of directional realism.

45. **THRESHOLD OF HEARING** The softest sound that a normal human ear can detect is called the threshold of hearing, and it varies in absolute strength at differernt frequencies. The threshold at 1,000 Hz is taken as a sound pressure of 20 micropascals, and this is the zero DECIBEL point of the SOUND PRESSURE LEVEL scale.

46. **TIMBRE** Timbre refers to the subjective quality or "tone color" of a sound. It is not related to PITCH or LOUDNESS. It is timbre which allows us to tell the difference between musical instruments. The timbre of a sound depends on many factors, including the strength and numbers of harmonics as well as the characteristics of the sound's transients.

47. **TONE** Tone refers to a signal which has a particular and usually steady PITCH. The simplest of tones are sine waves.

48. **TRANSDUCER** A transducer is a device which converts mechanical, magnetic or acoustic energy into electrical energy, or vice versa. Examples are microphones, phono cartridges and loudspeakers. In general, transducers are the weakest links in the chain and cause most of the distortion.

49. **UNITY GAIN** A device which does not attenuate or amplify a signal is said to have unity gain.

AUDIO TERMINOLOGY

59. **VU** VU is the abbreviation for volume unit, which is a measure of signal level on a meter with a decibel scale and accurately controlled rate of response to a signal. The reference power for the volume unit is delivered to a 600 OHM load when the voltage is .775 volts, rms. This is zero on the meter or 0 vu.

60. **WAVEFORM** The waveform of a SIGNAL is a graph of the instantaneous voltage versus time. The familiar sine wave is an example.

61. **WAVELENGTH** In a sound wave, the distance between two successive pressure maxima is called the wavelength, and it is equal to the speed of sound divided by the frequency. In standard conditions, sound travels at about 340 meters per second, so the wavelength of a 10,000 Hz sound is340/1000, or about 3.4 centimeters.

62. **WHITE NOISE** White noise is a special type of random noise where the energy content is the same at each frequency. Because of our ears' peculiar method of determining LOUDNESS of sounds (FLETCHER MUNSON EFFECT), white noise sounds as if it has more energy at high frequencies than at low.

RECOMENDED READING RESOURCES

1. **THE AUDIO DICTIONARY, 2nd. EDITION**
   GLEN D. WHITE
   A VIRTUAL MINI ENCLOPEDIA ON SOUND RECORDING, REINFORCEMENT AND ACOUSTICS THAT OFFERS MATH-FREE, PLAIN ENGLISH DISCUSSIONS OF MANY KEY TOPICS. 1991

2. **SOUND RECORDING HANDBOOK**
   JOHN WORAM
   THE DEFINITIVE ADVANCED REFERENCE ON ANALOG RECORDING TECHNOLOGY. IT BEGINS WITH A SOLID COURSE ON THE ESSENTIAL MATH AND PHYSICS OF AUDIO, THEN EXAMINES THE THEORY AND APPLICATIONS OF MICS, MONITORS, DELAY, REVERB, EQ, TAPE HEADS, NOISE REDUCTION CONSOLES AND TIMECODE. 1981

2. **MODERN RECORDING TECHNIQUES**
   HUBER & RUNSTEIN
   CLEAR DESCRIPTIONS OF EACH PIECE OF GEAR IN TODAY'S RECORDING STUDIOS, ALONG WITH CONCISE EXPLANATIONS OF THEIR FUNCTIONS. FULL CHAPTERS ON MICS, CONSOLES, TAPE DECKS AND SIGNAL PROCESSORS, PLUS COVERAGE OF MIDI, SYNCHRONIZATION, HARD DISC RECORDING, AUTOMATED MIXING, AND DIGITAL AUDIO. 1989

3. **RANDOM ACCESS AUDIO**
   DAVID MILES HUBER
   POPULAR NEW ENTRY-LEVEL GUIDE TO DIGITAL AUDIO, SAMPLING AND HARD DISK RECORDING. EXPLAINS THE FUNDEMENTAL CONCEPTS BEHIND DIGITAL AUDIO, AS WELL AS THE MAIN APPLICATIONS IN MUSIC, MULTIMEDIA AND BROADCAST. FILLED WITH CLEAR ILLUSTRATIONS AND PRACTICAL EXAMPLES. 1992

4. **MECHANICS OF SOUND RECORDING**
TONY ZAZA
A DETAILED LOOK AT THE TECHNOLOGY OF AUDIO FOR VIDEO AND FILM. FEATURES CHAPTERS ON MIXING, SOUND TRANSFER, EDITING, DUBBING AND VIDEO INTERFACE WITH A SPECIAL FOCUS ON LOCATION RECORDING. 1991

6. **SOUND FOR PICTURE-AN INSIDE LOOK AT AUDIO PRODUCTION IN FILM & TELEVISION**
EDITORS OF MIX
TAKES YOU BEHIND THE SCENES AS TOP HOLLYWOOD SOUND PROFESSIONALS REVEAL HOW DIALOG, SOUND EFFECTS AND MUSICAL SCORES ARE RECORDED, EDITED AND ASSEMBLED INTO SEAMLESS SOUNDTRACKS. FOCUSES ON BOTH THE EQUIPMENT USED AND THE PHILSOPHICAL SIDE OF SOUND DESIGN. 1993

7. **TIMECODE: A USER'S GUIDE**
JOHN RATCLIFF
AN ADVANCED, PRACTICAL HANDBOOK ON SMPTE/EBU TIMECODE AND ITS APPLICATION IN FILM, VIDEO AND AUDIO PRODUCTION. GOES BIT BY BIT THROUGH LTC AND VITC, MIDI SYNCHRONIZATION WITH HANS ON CHAPTERS ON STUDIO, LOCATION AND POST PRODUCTION TECHNIQUES. 1993

MAGAZINES

1. **MIX . MIX PUBLICATIONS, INC., BERKELEY, CALIF.**
INCLUDES SEMI-TECHNICAL ARTICLES ON MOST ASPECTS OF AUDIO AND STUDIO TECHNIQUES, SOME ARTICLES ON VIDEO AND REVIEWS OF STUDIO EQUIPMENT.

2. **VIDEOMAKER. VIDEOMAKER INC., CHICO, CALIF.**
BEGINNING GUIDE TO CURRENT TRENDS IN VIDEOMAKING INCLUDING ARTICLES ON AUDIO. VERY INFORMATIVE IN A NON-TECHNICAL WAY.

3. **RECORDING ENGINEER/PRODUCER (RE/P) HOLLYWOOD**
THIS SEMI-TECHNICAL MAGAZINE IS AIMED AT THE RECORDING STUDIO OPERATOR AND ENGINEER.

AUDIO SLANG

OOMY - Refers to a sound that has too much bass or low end, roughly centered around 100 Hz.

BOXEY - Refers to the area of the frequency bandwidth that makes a sound feel hollow or "inside a box". This usually unwanted sound can be cured by "rolling off" (lowing in volume) the bandwidth between 250 Hz and 500 Hz.

BRIGHT - Refers to a sound that has lot of treble or high end, roughly around 10kHz to 16kHz. Also used as a verb, ie. to "brighten" a sound means to add high end.

PRESENCE / SIBILANCE /INTELLIGIBILITY- Refers to area of the bandwidth that makes speech easier to understand. To enhance the clarity of the human voice, boost the upper mid range frequencies around 4kHz to 8kHz.

MUDDY - Refers to a sound that is dull and/or unclear. Such a sound is usually lacking in high end or has too much bass.

HISS - Refers to the very high frequencies above 12kHz that sound like how you pronounce the word "hiss". Low end analog tape recorders such as audio cassettes inherently have a lot of tape hiss.

WARM - Refers to a sound that is too "bright" or has too much high end and is lacking in the lower mid-range frequencies. Can be cured by rolling off the highs and/or boosting from around 200 to 500 Hz.

Glossary of Terms

A/D Converter
Analog-to-digital converter; a circuit that converts an analog signal to a digital signal. With an Audiomedia II card, DAT deck, or most other digital audio devices, each analog input *channel* is equipped with an A/D converter. A/D converters operate at particular *bit-rate resolutions*, and at particular *sampling rates*.

AES/EBU Digital Interface
Describes an industry standard for transferring digital audio between professional-quality audio devices. Stands for "Audio Engineering Society /European Broadcasters' Union." Most AES/EBU connections utilize an *XLR*-style 3-pin connector, which is able to carry two channels of digital audio (as opposed to one channel of analog audio). Most professional gear, including Digidesign's professional Pro Tools *audio interface*, is equipped to handle at least one AES/EBU *I/O*, and often more. See also *S/PDIF*.

AIFF
Audio Interchange File Format, an Apple audio file format that is supported by many Macintosh software applications. Session does not directly support this type of file in its sessions. If you wish to use an AIFF file within Session, you will first have to import it using the *Import Audio File* command. The AIFF format is best if you plan to use bounced audio in other Macintosh applications that do not support the Sound Designer II format.

Amplitude
A term used by recording engineers to describe relative levels. See *Signal Levels*.

Analog Audio
Audio which is transmitted, processed, amplified, or otherwise managed in the analog domain. Unlike digital audio, which represents audio in terms of discrete numbers, analog audio can be represented by continuously variable voltages. The signals between home stereo and recording components, for example, are usually analog audio.

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Audio Card
Refers generically to computer cards that process or generate sound. On the Mac, examples include Digidesign's Audiomedia II, Pro Tools, and Session 8 card; most Mac cards are of the NuBus standard, but an increasing number are PCI-bus standard. On the PC-compatible side, the most common card of all is Creative Lab's SoundBlaster; professional-grade cards include Digidesign's Session 8 PC; most PC cards utilize the ISA-bus standard, but as with the Mac, PCI-bus cards are becoming a de facto standard.

Audio File
A digital recording, as stored in digital form, typically on a hard disk. Audio files are stored in one of four primary file formats. Also see *AIFF, Sound Designer II File Format, Sound Resource File Format,* and *.WAV.*

Audio Interface
Among many *DAWs,* such as Digidesign's Pro Tools, the audio interface is a separate box that is attached with a special cable to the audio card. An audio interface typically has *analog* and *digital audio* inputs, and may be equipped with *level meters,* level controls, and other features.

Audio Post-Production
The process of adding dialog, music, sound effects, and other audio to video, film, or computer-based movie. The "post" reference comes from the fast that these elements are added once the film or video has already been shot.

Audio Region
See *Region.*

Audio Regions List
See *Regions List.*

Audition
[verb] A term that recording engineers use for listening to, or "monitoring," one or more *tracks* or audio *files.* Within Digidesign's Session program, for instance, you can audition files before you load them into the program; you can also audition individual *tracks* using the *solo* feature.

Autolocation
A feature that allows you to store and recall points of time into a recording system's memory. Session's autolocator function allows you to set *markers* at specific events. Individual locations can then be recalled, which automatically jumps the session's playback point to the marker's time.

Automation
A feature that allows you to record and playback changes in *fader* levels, *pan* controls, and *muting.* Session allows you to view and edit all fader and pan automation. During automation recording, all audio regions remain intact; only changes in automation are recorded.

Automation Record Enable
Within Session, an onscreen button that prepares a track to record *automation.* Similar to *record enable.*

Auxiliary Send
See *effects send.*

Balanced & Unbalanced Audio
Most professional +4dBu and microphone-level analog signals (see *signal levels*) utilize balanced audio, with balanced cables and balanced connectors. Digidesign's Pro Tools 882 audio interface, for instance, has balanced analog inputs and outputs. Balanced lines are typically less susceptible to noise (such as hum, radio frequency interference, and so forth) than unbalanced lines.

Here's how balancing works: A balanced cable has two conductors and a ground, and usually ends in either *XLR*-type 3-pin connectors or *TRS* 1/4" connectors. Both of the two conductors carry the audio signal. However, the "cold" or "negative" (–) conductor carries a signal that is 180° out-of-phase with the "hot" or "positive" (+) conductor. (See *phase.*) When the two conductors' signals are recombined at an audio device's input stage, a phase "inverter" reverses the cold conductor's phase, so that its audio signal is in-phase with the hot conductor. Why go to all this trouble? As audio travels through a balanced line, noise accumulates in both conductors. Then, when the two audio signals are recombined after the cold conductor's phase is inverted, two things happen: The noise signals are now out-of-phase with each other, so they cancel each other out. The audio signals, on the other hand, are added together, which produces an even stronger signal.

Most semi-pro and home stereo –10dBu analog lines are unbalanced, which means they have just a single positive (+) conductor and a ground. Digidesign's Audiomedia II card has unbalanced analog inputs and outputs, as do Apple's PowerPC and Quadra AV computers. Although noise can accumulate in the positive conductor, functionally speaking, these days, most unbalanced systems are capable of performing very quietly, with very low noise. Professional applications, however, demand balanced interconnections as a safeguard against noise.

If you need to connect a balanced output signal to an unbalanced input, the most reliable method is to use a "balancing transformer," which assures that the line stays balanced from output to input, and also helps to match *impedance.*

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Beat Markers
Session allows you to define both a tempo and a time signature to create beat markers within the *Edit window* that delineate the beat. This function is most useful for identifying location with a musical piece, and for *Grid* alignment of regions to the nearest beat.

Bit
The digital "alphabet" is composed of just two binary digits, or bits: "0" and "1". Digital circuits usually represent a 1 with a relatively "high" voltage, and 0 with a low voltage. Eight bits form a *byte.*

Bit-Rate Resolution
One of two main specifications that define digital audio quality. Along with *sample rate*, which defines an audio file's upper frequency limit, bit-rate resolution defines how precisely a sound's *dynamic range* is represented.

Remember that digital audio uses numbers to represent sound. The more numbers, the more precisely you can represent a sound. Consider first that analog sound — such as the sound we hear in the every day world around us — has a virtually infinite dynamic range of levels, from loud to quiet. Now let's imagine we wanted to capture this infinite range with a recording system that has a 2-bit recording system. With such a system, we could use four number combinations (2^2) in the following way:

11 = Loudest
10 = Medium Loud
01 = Quiet
00 = Silent

In other words, a two-bit audio file gives you just four possible levels — not even enough for a low-fidelity telephone answering machine. An 8-bit audio file, however (2^8) gives you a dynamic range of 256 possible levels, which isn't bad, but not nearly as good as the 65,536 levels that you get with 16-bit resolution (2^{16}). Practically speaking, 16-bit files have a much better *signal-to-noise ratio* than 8-bit files, which means they have much less audible noise. There's a rule-of-thumb you can use to figure out the approximate dynamic range of any particular bit rate: simply multiply the bit rate by 6. For instance, 8-bit audio has a dynamic range of "8x6" or 48dB; 16-bit audio has a dynamic range of "16x6" or 96dB

Professional applications call for 16-bit audio (some pros use 20-bit systems, such as Digidesign's ProMaster 20). Other resolutions, such as 8-bit audio, may be suitable for presentation or multimedia applications. As with sample rate, bit rate also has a bearing on the size of a file.

Bouncing
The process of combining two or more audio tracks into a single track. With Session, bouncing is much like mixdown — although instead of mixing to an external device (such as a DAT deck), you mix back to your system's hard disk. Bouncing is done to free up disk space or to free up *tracks*. The primary disadvantage of bouncing is that once you have performed the bounce, you can't change the relative levels or panning of the combined tracks. For this reason, the original "source" tracks should be deleted only once you are completely satisfied with the bounced track.

Byte
A "word" comprised of eight *bits*. A byte has 2^8, or 256 possible combinations, from "00000000" to "11111111".

Cardioid Microphone
A *cardioid*, or *unidirectional*, microphone is one that is sensitive to sound sources that emanate primarily from in front, of the microphone. Because of its design, a cardioid mike will reject most sound that emanates from beside the mike, and almost all sound that emanates from behind the mike. It's called "cardioid," by the way, because its response pattern resembled the shape of a heart. A cardioid microphone is the best all-around pattern for most music and dialogue recording, particularly when used outside of a professionally soundproofed studio, because it will be less likely to pick up unwanted background noise.

Channel
A channel is a discrete signal path. A mixing *console*, for instance, is a device that combines two or more channels. A channel is not necessarily the same as a *track* (although some people consider the two words synonymous). In Digidesign's terminology, channels refer to how many discrete audio inputs or outputs an *audio interface* has; tracks, on the other hand, are areas in the *Edit window* that allow you to playback simultaneous audio events.

Chorusing
A type of processing that many *digital effects processors* can perform. Chorusing adds a swirling, shimmering sound to your audio, similar to flanging but not quite as intense. Chorusing is actually a modulated type of *digital delay* setting, with a delay of about 15 to 30 milliseconds.

Clear
A Session function, similar to *Cut*, that removes a region from the *Edit window* or Audio Regions List. Unlike Cut, however, Clear does not store the removed region to the Macintosh's Clipboard memory.

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Clip Indicator
An light or other indicator that warns you that a level may have run out of *headroom*, and may be approaching *clipping*. Within Session, the clip LED functions as a clip indicator.

Clipping
Distortion that occurs when a signal is so high that is runs out of available *headroom*. The term comes from the shape of audio waveforms when viewed on an oscilloscope, or within the *Edit window*; the tops (and bottoms) of each waveform look as if they've been "clipped" off. Clipping of a digital signal generally sounds like a snapping or crunching sound (unlike analog clipping, which usually sounds "warmer," like the distorted sound of an electric guitar through a tube amplifier that is driven in clipping). While occasional clipping is tolerable (or even desirable) with analog signals, digital clipping must be avoided in all circumstances.

Compressor; Compressor/Limiter
A type of *signal processor* that effectively reduces a signal's *dynamic range*. A compressor can be used to make loud passages that exceed a preset "threshold" level sound less loud. Effectively, this makes quiet passages sound louder and loud passages sound quieter. A compressor is typically used to "smooth out" dynamically uneven tracks — such as a vocal track where the vocalist has not kept a very consistent level. It can also be used with a bass guitar or other instrument to enhance the apparent sustain, by making the quiet decaying portions sound louder. Extreme compression forms a type of *limiter*, which sets a maximum threshold level that cannot be exceeded. Limiting is useful to prevent *clipping*.

Connectors
Refers to the many different types of input and output jacks, and cable plugs, that are used in audio recording. The most popular connectors are *Quarter-inch*, *Eighth-inch*, *XLR*, *RCA*, and *TRS*.

Condenser Microphone
A type of microphone that uses a charged, or polarized, diaphragm to capture sound. Condenser mikes can be externally powered from a mixer or microphone preamp; this type of power is called phantom power, because it travels "unnoticed" through the microphone cable, without affecting the audio signal. Phantom power is usually 48 volts DC, and requires a professional-style balanced microphone and microphone input (or microphone *preamplifier*). Some condenser mikes are internally powered, using a small battery (usually AA or 9v).

An electret microphone is similar to a condenser, except that its diaphragm is permanently charged, and doesn't require power; however, most electret microphones still require phantom or internal battery power to drive their preamplifiers — which raise the relatively weak signal from the capsule up to a microphone level.

Copy
A Session function that allows you to copy a *region* (or regions) in the *Edit window*, much like the copy function of a word processor. Copied regions are held in the Macintosh's Clipboard memory in order to be *pasted*, until something new is copied or *cut* into the Clipboard.

Crossfade
[verb] To reduce the playback level of one portion of audio as you raise another portion — much like a dissolve between scenes in video or film. Crossfades are performed using either an external *mixing console*, or (with Session or most other *DAWs*) the onscreen *faders* within the *Mix* window, or a "crossfade" function.

Crossfade
[noun] In Session, the crossfade function allows you to *fade out* from one region as you *fade in* to another region. Crossfade types and duration are user selectable from within the *Edit window*. As with *fades*, portions of audio for which the fade function has been applied are stored in Session's "Fades" folder.

Cut
A Session function that allows you to remove a *region* (or regions) in the *Edit window*, much like the cut function of a word processor. Cut regions are held in the Macintosh's Clipboard memory in order to be pasted, until something new is copied or cut into the Clipboard.

D/A Converter
Digital-to-analog converter, a circuit that converts a digital signal to analog. With an Audiomedia II card, DAT deck, or most other digital audio devices, each analog output *channel* is equipped with an D/A converter, much as each input channel is equipped with an A/D converter.

DAE
See *Digidesign Audio Engine*.

DAT
Stands for Digital Audio Tape, a type of cassette that uses 4mm tape. A DAT deck a common type of stereo recording deck used in professional studios. DAT is also used as a SCSI backup drive, capable of holding up to 8GB of data. Audio DAT decks and DAT backup drives are different devices, even though they use the same type of cassette.

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DAW
See *Digital Audio Workstation.*

Decibel
The primary unit for measuring audio levels; also spelled deciBel, for Alexander Graham Bell. There are a variety of different decibel scales, but for our purposes, the primary scale that allows us to measure different devices on the same level "playing field" is known as the *dBu*, or *dBv* (not the same as *dBV*). Also see *Signal Levels.*

Defragmentation
As data is written and erased from a hard disk (or a hard disk partition), gaps form between different "chunks" of data. These gaps can slow down the hard disk's *seek-time* speed, as the drive's heads have to seek here and there to find data, and this can impede audio playback performance. Consequently, it's important to "defragment" the drive frequently, using a program such as Symantec's Norton Utilities. To play it safe, Digidesign recommends you defragment frequently. Alternately, erasing *everything* on the hard disk regularly allows you to begin writing contiguous data, that will be free of seek-time troubles.

Delay
A type of audio processing that many *digital effects processors* and certain types of software can perform. Delay is the process of storing portions of an audio signal briefly in memory, and then playing them back. With the typical digital effects processor, delay times can be adjusted from about 1/2000 of a second (0.5 milliseconds) to one or two seconds (1000 to 2000ms), or longer. Delay can be used to create an *echo* or a *doubling* effect. When the delay's signal is "modulated" — that is, its time is varied slightly at a user-definable rate — modulated delay effects such as *chorusing* and *flanging* can be created.

Destination
Something that receives an audio signal. See *Sources & Destinations.*

Digidesign Audio Engine (DAE)
Digidesign's real-time operating system for its digital recording systems. When you install your Session Software, DAE is automatically installed inside your System folder into a DAE folder. In the same way that the Macintosh System software provides the foundation for Macintosh software applications, DAE provides the core functionality of hard disk recording, digital signal processing, mix automation, and MIDI required by Session 8 and other Digidesign products. Because DAE is an application itself, separate from Session, it supplies these same capabilities to products such as Studio Vision™, Logic Audio™, Digital Performer™, and Cubase Audio™ which utilize Digidesign DSP resources to record and play digital audio.

Digital Audio
Audio which is transmitted, processed, amplified, or otherwise managed in the digital domain. Digital audio uses numbers — *bits* comprised of "0" or "1" — to represent sound. Digital audio requires an *A/D converter* to convert sound from analog into digital, and a *D/A converter* to convert it back to analog. Digital audio quality is described primarily by two specifications: the *bit-rate resolution*, and the *sampling rate.*

Digital Audio Workstation (DAW)
The more common name for a *hard disk recording* system, that can be used to record, edit, and mix audio. Certain types of audio software, such as Digidesign's Session, can also be used to turn a computer into a DAW. Most DAWs use a personal computer as a "host" (such a Digidesign's Session) and may also require an additional *audio card* and *audio interface* (such as Digidesign's Pro Tools).

Digital Delay
See *delay.*

Digital Effects Processor
A device or type of software that applies various types of effects processing — such as *reverberation, chorusing, doubling*, etc. — to an audio *source.* Uses *DSP* technology to perform its functions.

Distortion
Occurs when usually unwanted components are added to a sound, often as a result of *clipping.* (Some types of distortion are desirable, such as the "warm" clipped sound of a electric guitar amplifier, but most are not.) *Analog audio* distortion is most common, but digital equipment can be susceptible to distortion, especially in the *A/D* and *D/A converters.* Most equipment specifications include two common types of analog distortion: *harmonic distortion* and *intermodulation distortion.*

Doubling
An effect similar to an *echo*, but with only a single "slapback" of sound. The result is something that sounds as if there are two of the original source. In Session, a track or region can be doubled by copying it and pasting the copy onto a new track, and then "slipping" the new track roughly 1/20 to 1/2 of a second (50 to 500 milliseconds) behind in time. Using a digital effects processor, a track can be doubled with a delay setting of 50 to 500ms.

Drop-Frame
Refers to a variance of *SMPTE/EBU timecode* which omits two frames every minute except for every tenth minute. Serves to compensate for the fact that 29.97 frame-rate

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timecode runs "slow" by 3.6 seconds every hour compared to a 30 frame-per-second timing reference; that is, "one hour" of 29.97 non-drop timecode is equal to one hour and 3.6 seconds of real time, due to the fact that the slower frame rate does not match "wall clock." Dropping — or more descriptively, skipping — frames allows this format to "jump ahead," in order to maintain a true relationship between the timecode and "real time," so that one hour of 29.97 drop-fame does indeed last exactly one hour of real time.

DSP
Digital Signal Processing. In audio terms, DSP refers to manipulation of digital audio — everything from *reverberation* to changes in level.

Duplicate
A Session function that allows you to duplicate a *region* in the *Edit Window*. Unlike the *copy* function — where the region is held in the Macintosh's Clipboard memory until it is *pasted* — duplicate automatically places a copy of the selected region adjacent to it. Duplicate is useful for repeating regions, such as a repeated musical passage, sound effect, and so forth.

Dynamic Range
The variation between a signal's quiet and loud passages, measured in decibels. Consider a symphony orchestra, which is capable of great dynamic variation. At its loudest, the typical orchestra may be able to produce roughly 110 to 115dB (using the "acoustic" dBA scale for decibels, the same sort of scale that's used for measuring the noise of a jet aircraft). At its quietest moments, perhaps the most pianissimo flute passage, the orchestra may measure only 50dB. (Any quieter, and the flute would probably be drowned out by the "ambient" noise of the concert hall, caused mostly by the hall's air conditioning and other background noise.) Given this span, we could say this orchestra has a dynamic range of roughly 60 to 65dB; if we include the portions when the orchestra is silent, and all we're left with is the ambient sound of the hall, the dynamic range could then be over 80dB. Even this, however, is well within the capabilities of 16-bit audio — which has a dynamic range of 96dB.

Dynamic range is similar to, but not the same as, *signal-to-noise ratio*. Also see *bit-rate resolution*.

Echo
A type of audio processing that many *digital effects processors* and certain types of software can perform, and a variation of *delay*. An echo is one or more distinctly audible repetitions of a portion of sound. Unlike delay, however, which can be as "short" as 1/2000 of a second (0.5ms), echo is usually no shorter than 1/20 of a second (50ms), since times less than that no longer sound like a distinct repetition. Multiple echoes are achieved using a *feedback* control, which sends the first echo (and if desired, subsequent ones) back into the delay-generating process.

Edit Window
In Session Software, the Edit window is where *regions* and *tracks* can be arranged, for moving sound about in time; for copying, pasting, duplicating, or cutting regions; and more. In the Edit window, tracks appear as graphic *waveforms*. As in the *Mix* window, each track has controls for audio record-enable, automation record-enable, and mute/solo.

The Edit, *Mix*, and *Transport* windows are Session's three main control environments.

Effects Send
Mixing consoles (and some DAWs) include effects sends, which allow you to route the signal from individual channels to an external effects device (such as a *digital effects processor*). Digidesign's Session does not include a true effects send, although a means of adding external effects is described in Chapter F, *Mixing*.

Eighth-Inch (1/8") Connector
Also called a miniphone connector, miniplug connector, just a mini connector. This relatively small connector is used most commonly as an inexpensive input or output connector for analog audio. For instance, a stereo 1/8" connector is commonly used as a headphone output jack on CD-ROM players, portable stereos, and inexpensive sound cards, or as a microphone or line level input jack on sound cards. Apple PowerPC- and Quadra AV-series computers use 1/8" connectors for their analog audio inputs and outputs.

Electret Microphone
See *Condenser Microphone*.

Equalization (EQ); Equalizer (EQ)
The process of cutting or boosting one or more areas of a particular sound's *frequency range*. EQ is generally used as a tone-shaping tool — for instance, to increase bass, decrease treble, or bring out a particular "brightness" or other tonal characteristic. EQ works by boosting or cutting specific frequency ranges. The bass and treble controls of a home or car stereo are what is called a 2-band equalizer; add a mid-range control and you have a 3-band EQ. (Session provides two bands of EQ per track.)

Session provides five types of EQs. Each EQ has controls for selecting the EQ type, Gain (the amount the frequencies are cut or boosted), Frequency (the specific frequency to be cut or boosted), and a Bypass button for disabling the EQ. In the case of the parametric EQ, an EQ Bandwidth selector is also included (for choosing how wide a range of frequencies will be affected). Also see *Low-Shelf EQ; High-Shelf EQ; Parametric EQ; High-Pass EQ;* and *Low-Pass EQ*.

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A graphic EQ is another common type of EQ. "Outboard" graphic EQ *signal processors* usually have five to 30 bands of EQ, each with a fixed frequency. They're called "graphic" because they usually use slider-type controls, and when you view them as a whole you get a graphic representation of how the sound's frequency range is being affected by the processor.

Expander

A type of *signal processor* that effectively increases a signal's *dynamic range*; in some ways the opposite of a compressor. An expander can be used to make quiet passages that drop below a preset "threshold" level sound quieter, and is an effective means of reducing unwanted sounds, such as tape hiss, background rustling of papers, and so forth. An expander set to operate in an extreme mode is also called *gate*, which effectively silences signals that drop below the preset threshold level.

Fade

[verb] Normally synonymous with "fade-out" — to reduce the playback level of one or more tracks. Fading is performed using either an external *mixing console*, or (with Session or most other *DAWs*) the onscreen *faders* within the *Mix* window, or Session's "fade" function.

Fade; Fade-In; Fade-Out

[noun] A section of audio where the level has been increased or reduced. A "fade-in" is where the level is raised from inaudibility; a "fade-out" is where the level has been reduced to inaudibility. With Session, fades can be accomplished using either *automation*, or using the "Fade" feature, where a region or portion of region is selected, and raised or reduced in level using one of the program's preset fades. As with *crossfades*, portions of audio for which the fade function has been applied are stored in Session's "Fades" folder.

Fader

Most commonly, a type of *potentiometer* found on a *mixing console* that controls individual *channel* levels. Faders are usually "linear" pots — that is, instead of working in a rotary fashion, they slide in a straight line. Digidesign's Session, and most other DAWs, have onscreen "virtual" faders, that are manipulated either be a mouse or an external *MIDI controller*. Faders are usually calibrated to indicate the amount of boost or cut in decibels. With most systems, the lowest possible level is usually indicated as "– ∞", as in "Infinite cut." Highest levels are typically shown as "+3dB" to "+10". The nominal level, sometimes referred to as "unity gain," is represented as "0dB" — which is not the same as "no level at all." Rather, a fader setting of 0dB is one which neither boosts nor cuts the audio level.

Feedback

The most common form of feedback occurs when a "live" microphone is brought near a loudspeaker or headphone: The mike picks up the loudspeaker, the signal is then re-amplified through the speaker, which gets picked up again by the mike — and on it goes, forming a squealing feedback "loop." Consequently, whenever working with a microphone in the same room as monitor speakers, it's important that the microphone's monitor level signal be kept low or off. When working with headphones, "closed ear" or "semi-ported" headphones should be used, which form a relatively tight seal around one's ears.

"Electronic" feedback is also a danger, especially with a *mixing console*, where the output of a device might accidentally be routed to the input of the same device.

Filter

Usually refers to a type of equalization designed to remove one or more or a sound source's frequencies. See *High-End Filter* and *Low-End Filter*.

Flanging

A type of processing that many *digital effects processors* can create. A "swirling"-type effect, similar to *chorusing* but more intense. Flanging is also a modulated type of *digital delay* setting, with a delay of about 0.5 to 15 milliseconds.

Frequency

A means of describing how frequently a particular tone or sound source induces periodic cycles. We can hear these cycles as sound waves if they exist in the natural world — or if they have been amplified through a loudspeaker or headphones — and are within our *frequency range* of hearing. Frequency is usually measured in Hertz (Hz), the common term for cycles per second (cps).

Frequency Range

A range of frequencies, usually as they apply to a particular sound source, and usually expressed in Hertz. The frequency range of human hearing is generally considered 20Hz to 20,000Hz (20kHz) — but practically speaking, most adults' hearing is limited to an upper-end response of 12kHz to 16kHz.

Frequency Response

Most frequently refers to a specification that describes a particular component's ability to reproduce the *frequency range* of human hearing. Ideal response, in most circumstances, is so-called "flat" response, where all frequencies from 20Hz (or lower) to 20kHz (or higher) are reproduced with equal *amplitude*. It's called flat because the a chart of the frequencies on the horizontal axis, with the "deviation" on the vertical

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axis, would look like a flat line. Depending upon the standards applied, a deviation of anywhere from 0.1dB to 3dB can still be considered flat. For instance, flat frequency response with a deviation of 3dB would be indicated as "20Hz – 20kHz ±3dB".

Gain
Boost (amplification) or a signal. See *Signal Levels*.

Gate
A type of *signal processor*. A gate is actually an *expander* set to operate in an extreme mode, so that signals which drop below a preset "threshold" level are effectively silenced. Whether used as an outboard processor or as a software program, a gate is a good way to eliminate occasional unwanted background sounds, such as breath noise, paper rustling, and so forth.

GB
Abbreviation for *Gigabyte*.

Gigabyte
One billion, or one thousand million, bytes (actually 1024 megabytes, or 1,073,741,824 bytes). A 1GB *hard disk* is generally a good minimum size for digital audio production purposes. Abbreviated GB or sometimes GByte.

Grabber
A tool within Session Software. The Grabber lets you move or rearrange *regions* simply by dragging them to a new location in a track.

Grid
Session's Grid mode is a means of aligning *regions* in tracks to the nearest unit of a user-defined value. When Grid mode is on, the Selector will "snap to" the currently selected grid mode value, and regions placed in the Edit window will also line up with the selected value. Grid mode is similar to the "quantizing" of note start times in MIDI sequencers, or the "Snap to Ruler" function of many page layout programs.

Grid Mode provides three types of external alignment references: alignment to *time bases*, alignment to the beginning and end of other *regions*, and alignment to *beat markers*. These external references create invisible "snap points" to which the start points of regions are attracted.

Grouping
The linking of two or more *faders*, in order to move them as a group; a feature common to automated *mixing consoles*, and many *DAWs*. Grouping, for instance, allows you to control the overall level of several tracks of background vocals — by moving just one fader, while retaining the same relative level of the different tracks. Session, for in-

stance, let's you create up to four different groups — not only of level faders, but also of pan controls. A collection of grouped faders can also be thought of as a "submix," or a "subgroup."

Hard Disk (Hard Drive)
A data storage device capable of holding relatively large amounts of data; named because the magnetic platter(s) upon which data is written are physically hard, as compared to a floppy disc, which uses a pliable, tape-like magnetic surface. For digital audio recording, typical one or more hard disks of 1 *gigabyte* or larger are in order.

Hard Disk Recording
Digital recording that records and plays back audio to and from a hard disk, as opposed to tape. A hard disk recording system is more commonly called a *digital audio workstation* (DAW).

Harmonic Distortion
A common type of *distortion*; sometimes called "THD," for Total Harmonic Distortion. Refers to distortion that contains new frequencies that are harmonic multiples of the original sound's frequency. Generally, most people can detect harmonic distortion when it reaches a level of 0.5 to 1% of the sound's complete harmonic content, although most recording equipment has THD specifications of less than 0.1%.

Because of the essentially musical relationship between the original sound and its harmonic distortion components, this type of distortion is sometimes desirable, and is certainly less objectionable than *intermodulation distortion*. For instance, the "warm"-sounding distortion of a tube guitar amplifier is largely harmonic distortion.

Headroom
The amount of remaining gain available for a given signal before the onset of unacceptable *distortion*. Practically all analog audio devices (and all digital devices with *A/D converters*) have a specified amount of headroom for their inputs; most devices have roughly 10dB to 24dB of available headroom above their nominal input level (typically –10dBu or +4dBu). See *Signal Levels*.

Hertz (Hz)
The unit of measurement for frequency; also the same as cycles per second (cps). A thousand Hertz is referred to a kiloHertz, of kHz. Also see *Frequency*, *Frequency Range*, and *Frequency Response*.

High-End Filter
A type of *equalization* that reduces high frequencies. See *Low-Pass Equalizer*.

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High-Pass Filter (EQ)
Effectively removes all frequencies below the selected frequency setting (called a "hinge" frequency), while allowing all other frequencies to pass through. Confusingly, also called a "low-end filter," since it filters out low frequencies, while letting higher frequencies pass.

High-Shelf Equalizer (EQ)
Boosts or cuts frequencies at or above the selected frequency setting (called a "shelf" frequency). Also see *Equalization*.

Hypercardioid Microphone
A more "extreme" version of a cardioid microphone, with even less sensitivity to sounds emanating from the side. This can be a good choice when background noise is a major problem, but it has two major trade-off: 1) The mike has to point exactly at the sound source — if it or the sound source moves even a little bit, levels will drop dramatically; and 2) a hypercardioid design causes certain frequencies to be canceled, so the overall quality of sound is less good than with an equivalent cardioid mike. Also called "supercardioid."

IMD
See *Intermodulation Distortion*.

Impedance
A type of electronic "resistance" to signal flow, measured in "Ohms" (Ω). Impedance is also sometimes referred to as "Z" — as in low-Z (low impedance) or high-Z (high impedance).

The details of impedance are quite complicated, but there are several key things to know about it: 1) most *balanced* microphones and other professional-quality gear have low-impedance output signals, usually about 100Ω to 600Ω; 2) most musical instruments, home hi-fi stuff, and other *unbalanced* "semi-pro" gear (including many less expensive *audio cards*) have high-impedance output signals; and 3) most gear — whether home hi-fi, semi-pro, or professional — has high-impedance line inputs.

Here's how this knowledge helps you: Low-impedance output signals are designed to maintain signal quality over long cable lengths, even up to a hundred meters (330 feet) or longer. High-impedance output signals, however, tend to loose high-end *frequency response* if the cabling is much more than 7 meters (23 feet), and perform poorly if the cable length approaches 15 meters (50 feet) or more. Consequently, in the typical project studio, high-impedance outputs are not necessarily a liability, although because they are almost always unbalanced, they may be more susceptible to noise than a low-impedance output (which is often, but not always, balanced).

Regarding line input impedance, equipment used to be designed such that low-impedance outputs were always to be connected to low-impedance inputs, and that high-impedance outputs were always to be connected to high-impedance inputs. As mentioned above, however, most current gear utilizes high-impedance inputs. To figure out whether or not one type of gear will connect properly to another, remember this rule of thumb: *Low into high will fly; high into low won't go.* In other words, low-impedance output signals can be plugged into either low or high-impedance inputs, whereas high-impedance output signals require high-impedance inputs.

Intermodulation Distortion (IMD)
Unlike *harmonic distortion*, which in some situations is considered "warm" and desirable, intermodulation distortion is a particularly nasty-sounding form of *distortion*. The main difference is that whereas the components harmonic distortion have a "musical" relationship to one another, the components of IMD have a dissonant, "enharmonic" relationship. This also explains why our ears are more sensitive to IMD, and are generally able to detect as little as 0.01% to 0.1% of it, in relation to the non-distorted signal. Most quality gear has a spec of less than 0.005% IMD.

I/O
Abbreviation for *Input/Output*. "I/Os" is an abbreviation for Inputs/Outputs.

Jack
A "female"-style connector, typically an input. *Plugs* connect to jacks.

kB
Abbreviation for *Kilobyte*.

kHz
Abbreviation for *kiloHertz*.

Kilobyte
One-thousand *bytes* (actually 1,024 bytes). Abbreviated kB or sometimes kByte.

KiloHertz
1,000 *Hertz*. Also see *Frequency*, *Frequency Range*, and *Frequency Response*.

Level Meters
See *VU; VU Meters*

Limiter
See *Compressor/Limiter*.

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Line Levels
See *Signal Levels.*

Low-End Filter
A type of *equalization* that reduces low frequencies. See *High-Pass Equalizer.*

Low-Pass Filter (EQ)
Effectively removes all frequencies above the selected frequency setting (called the "hinge" frequency) while allowing remaining frequencies to pass through. Confusingly, also called a "high-end filter," since it filters out high frequencies, while letting lower frequencies pass.

Low-Shelf Equalizer (EQ)
Boosts or cuts frequencies at or below the selected frequency setting (called a "shelf" frequency). Also see *Equalization.*

Marker
A feature within Session that allows you to "mark" or locate a specific point in time during the *session.*

Mastering
1) The process of *mixdown.* 2) The final stage of preparing a audio for replication as a compact disc, CD-ROM disc, vinyl record, or analog tape. A "mastering house" is a facility that specializes in this process.

Mastering Deck
A 2-track (*stereo*) audio recorder, typically to which a *multitrack recording* is mixed. Common mastering decks include a *DAT* deck, an open-reel analog tape machine, or even an analog cassette recorder.

MB
Common abbreviation for *megabyte.*

Megabyte
One million bytes (actually 1024 kilobytes, or 1,048,576 bytes). Abbreviated as MB or sometimes MByte.

Microphone
See *cardioid, hypercardioid,* and *omnidirectional* microphone.

Microphone Preamplifier
See *Preamplifier*

MIDI
The Musical Instrument Digital Interface standard, which is a language that musical instruments, recording equipment, and computers can use to communicate information back and forth. MIDI uses 5-pin "DIN" connectors. Computer-based DAWs, such as Digidesign's Pro Tools or Session software, use MIDI to communicate time code information, Mix window fader levels, and much more.

MIDI Channel
MIDI utilizes 16 different channels. Much like a television, a MIDI device typically is set to receive (or transmit) information on only one channel at a time — although most MIDI devices can actually handle multiple channels of information at once.

MIDI Controller
An external device that uses MIDI information to manipulate a MIDI device. For instance, Digidesign's R1 or JL Cooper's FaderMaster have physical *faders.* Using MIDI, these devices can generate "continuous controller" information that can be used to adjust the levels of on-screen Mix window faders in Session. Similarly, "switch controller" information can be used so that physical buttons can control on-screen buttons, such as *solo, mute,* and so forth.

MIDI Timecode
See *MTC.*

Mini Connector
See *Eighth-Inch Connector*

Mixing Console
A device that allows you to sum together the signals from two or more *channels* of audio. Most mixing consoles have between four and 32 channels. Typically, each channel has two or more bands of *equalization,* plus several *effects sends,* a *pan control,* a *fader,* microphone and *line-level* inputs, *level meters,* and more.

Mix Window
In Session, the Mix window has controls that perform most of the functions of a true mixing console, only onscreen instead of an as an outboard piece of equipment. The Mix window is used to record tracks and adjust *monitor* and *mixdown* levels, *equalization,* and more. Each of Session's tracks has its own controls for *monitor level, record-enable, pan, solo, mute, bounce* to/from, and more.

The *Edit,* Mix, and *Transport* windows are Session's three main control environments.

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Mixdown

The last stage of an audio recording project, where, typically, multiple tracks of audio are mixed together to a *stereo DAT* cassette, an analog cassette, an open-reel tape, or a *audio file* on a *hard disk*. During mixdown, final decisions are made regarding the relative levels of different tracks, *EQ* settings, *automation*, *panning*, any *digital effects processing*, and more.

Monaural (Mono)

Sound with just a single channel of information. Even if played back on a stereo system with two speakers, a mono tape, track, or audio file will playback the same information through both speakers. (Assuming the pan and balance controls are "centered.") With Session (or any DAW), mono audio files take up half as much hard disk space as equivalent stereo audio files.

Monitor Level

Refers to the level at which one listens to one or more tracks, or a complete mix. On a *mixing console*, the monitor level adjusts the level sent to headphones, or to a *power amp*, which in turn drives *monitor speakers*.

In *DAWs* such as Session, monitor level controls can be used to adjust the level at which you listen to different tracks. They have no bearing upon *record level*. The collection of different control settings forms a "monitor mix."

Monitor Mix

See *Monitor Level*.

Monitor Speakers

Loudspeakers designed for recording studio use. Professional studio monitors are designed to be as "accurate" as possible, with as flat a frequency *response* as possible. This contrasts to many home stereo speakers — and almost all "multimedia" speakers — which often are designed to emphasize bass and treble response. Many monitor speakers are also *shielded*.

MTC

MIDI Timecode. An industry standard for transmitting the MIDI equivalent of *SMPTE/EBU timecode* between devices. While not the same as SMPTE timecode (it's a digital, rather than analog signal, for one), a SMPTE-to-MTC converter can be used to bridge the gap. MTC can follow all the frame-per-second rates of SMPTE timecode.

Multitrack Recording

The process of recording different passages of audio or music on more than one discrete track. Multitrack recording allows for "overdubbing," where a new track or part can be added to a track that has already been recorded. There are various types of multitrack recording systems, from tape recorders (with any where from four to 48 tracks), to *MIDI sequencers* (which usually have dozens of available tracks), to *DAWs* (which offer anywhere from two to 96 or more tracks).

Some DAWs (though not Session) differentiate between the number of available "simultaneous" tracks and "virtual" tracks. Simultaneous tracks refer to how many tracks can be heard playing at once — which is a limitation of the number of *voices* that the DAW has available. Virtual tracks, however, refer to those tracks that can assembled within an *Edit window*, but which are not necessarily heard during playback. Virtual tracks allow you to keep a number of different "takes," or edits of audio onscreen, and to assign them an available voice when you wish to hear them.

With Session, every available track — which depends upon the type of computer or *audio card* you're using — is a true track, with its own voice.

Mute

A common *mixing console* feature that is used to remove one or more *channels* from the mix, so that they are no longer heard. Digidesign's Session includes a mute function for each of its tracks.

Noise Gate

See *Gate*.

Nondestructive Editing

Editing that leaves *audio files* intact. As you edit audio within Session, all you are editing are the *regions*, or "pointers," to audio files that are stored on the hard disk. This is called "nondestructive" editing because the original audio files are always left intact (unless you choose to delete them, or overwrite them with new files that are based upon edited regions). This way, you can edit to your heart's delight, and always return to the original audio files if necessary.

Nondestructive Recording

Recording that leaves previously recorded *takes* intact. With Session, as you record each new take, previous takes are left on the *hard disk* as separate *audio files*, allowing you to return to a previous take (or portions of a previous take) whenever you so desire — unless you choose to delete your previous takes. The one disadvantage of nondestructive recording is that you may need lots of hard disk space, particularly if your takes are long and numerous.

Non-Drop-Frame

Timecode in which frames haven't been "dropped." The standard format outside of color video production or post-production is typically 30 FPS non-drop.

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Omnidirectional Microphone
A microphone that is equally sensitive to sound from almost all directions. An omnidirectional mike is a good choice when you only have one mike and you want to pick up sound from a variety of directions, and (such as roundtable discussion). This benefit is also the main drawback of an omnidirectional mike, since unwanted background noises — from cars to dogs to an overhead airplane — can be picked up easily.

OMS
Open MIDI System, developed by Opcode. OMS handles "background" MIDI issues, such as tempo and *synchronization*, and shares the information between OMS-compatible programs, such as Session. Using OMS, for instance, Session can communicate with and synchronize to such full-featured sequencing software packages as Opcode's Vision™, Passport's MasterTracks Pro™, Mark of the Unicorn's Performer™, Emagic's Logic™ and Steinberg's Cubase™. For instance, OMS allows you to record Session tracks while you listen to a previously recorded MIDI sequence.

Response Pattern
Also known as *polar response pattern*. Describes the shape of a microphone's sensitivity to sound. The three main patterns are *cardioid* (unidirectional), *hypercardioid* (supercardioid), and *omnidirectional*.

Pan Control
A mixer-style control which allows you to place a particular track or sound within the *stereo field*, from left to center to right. Session includes pan controls on each channel within the Mixer window. Pan stands for "panorama." Often referred to as a pan "pot" — see *Potentiometer*.

Parametric Equalization (EQ)
Boosts or cuts only those frequencies centered around a selected frequency setting. In Session, a pop-up menu allows you to select the width (from 1/3 to 3 octaves) of the parametric EQ. This determines the width of the filter's overall slope—from a broad "bell" shape to a narrow "notch." Also see *Equalization*.

Paste
A Session function that allows you to place regions that have been *cut* or *copied* back into a track in the *Edit window*. Cut and copied regions are held in the Macintosh's Clipboard memory until they are pasted, or until something new is copied or cut into the Clipboard.

Patch Cord
A cable that links an input and an output; typically a short cable, as would be used with a *patchbay*.

Patchbay
Essentially, a "switchboard" of inputs and outputs; a common studio accessory. Typically, device outputs and inputs are all connected to the rear of one or more patchbays. The rear-panel connectors also show up on the front of the patchbay, where the various inputs and outputs can be labeled. This way, using *patch cords*, devices can be quickly and easily connected together, without having to scramble around behind different pieces of gear. The most convenient types of patchbays for general use utilize *1/4" unbalanced* (or *TRS balanced*) jacks on both the front and rear panels.

Peak Indicator
An indicator light designed to warn of the possibility of *clipping*, which illuminates as a device's input reaches a preset degree of *headroom*. Usually part of a *VU meter* (on a mixing console).

Phantom Power
A type of DC power (usually 48 volts) that is used to power many condenser and some electret microphones. See *condenser microphone* for more information.

Phase
Refers to the relationship of audio signals, in terms of time. If two identical audio signals originate from the same source at the same time, they will travel through air "in-phase," so that the sound wave "peaks" and "troughs" rise and fall at the same time. When two in-phase signals are combined, their signal level is essentially doubled.

If, however, the two sounds were "out-of-phase," so that as one sound wave peaked the other troughed, when combined, they would cancel each other out — so that neither signal could be heard. *Balanced* audio uses principals of phase to help reduce noise.

Phone Connector
Another name for *Quarter-Inch Connector*.; comes from the fact that these connectors were used for when telephone companies still operated manual switchboards.

Phono Connector
Another name for *RCA Connector*; come from "phonograph."

Playlist
Session Software provides you with between 4 to 16 tracks of recording and playback. However, Session has the unique ability to load or unload what is on each of these

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tracks. This gives you the ability to record or create more alternative takes than you have actual physical tracks. These alternative takes or arrangements are called *playlists*. Session Software allows you to create an almost unlimited number of playlists and load and unload them as you wish.

Plug
A "male"-shaped connector, typically found at the end of a cable. Plugs plug into *jacks*.

Potentiometer (Pot)
A type of control, commonly found on mixing consoles, outboard effects processors, and so forth, that varies resistance in order to control pan, level, EQ, etc. A *fader* is a common type of potentiometer. Usually abbreviated as a "pot."

Power Amplifier
In the recording studio, a power amplifier uses high current to raise line-level signals — usually from a *mixing console, audio card*, or *audio interface* — up to a level suitable to *monitor speakers*. Most studio power amps are *stereo*, or 2-channel, devices. In most personal studio applications, a home stereo receiver or power amplifier can substitute for a professional amp. Suitable power ranges run from about 30 watts per channel (for relatively quiet listening situations) to 100 watts or more per channel. While it is desirable to match a power amp's output to your monitor speakers' rated power, it is actually better to somewhat overpower your speakers than underpower them, since too little power can introduce momentary clipping, which can damage monitor speakers more readily than momentary overpowering.

Preamplifier
In recording studio terminology, a circuit designed to boost relatively low *signal levels*, such as a microphone output, up to standard line levels of -10dBu or +4dBu. Practically all *mixing consoles* include preamplifier stages in each of their input *channels*.

Quarter-Inch (1/4") Connector
A very common type of connector found in the project studio, usually used for *unbalanced analog audio* connections. A variation, called the *TRS* connector, provides for either *balanced* or *stereo* operation. Also called a *phone connector*.

QuickTime
Apple's system extension for control of time-based events, such as digitized video "movies" and digitized sound. Originally developed for the Macintosh platform, QuickTime is also available for Windows. Session is able to import QuickTime movies to allow you to assemble audio and music events in *synchronization* with the movie. QuickTime movies can be converted to or from ordinary video with a "video capture card," or with an "A/V"-equipped computer, such as the Quadra 840AV.

RCA Connector
A type of connector used most commonly with home stereo setups, but also found commonly with multimedia and project studio equipment. The RCA connector's historical application has been for *unbalanced analog audio*, but it is also used commonly for "composite" video and also *S/PDIF digital audio* connections. Also called *Phono Connector*.

Record Enable
Within Session, an onscreen button that prepares a track to be recorded. Recording begins once the record button in the *Transport window* has been selected and the play button is engaged.

Record Level
Within Session, the level at which individual tracks are recorded. Session itself does not have a means of controlling input and record levels (they're determined either by the mixer or other sound source that is feeding your *Audiomedia II card* or Power Mac computer). During recording, it's important to use Session's *level meters* to help you adjust your mixer's (or other sound source's) output level, to avoid clipping.

Region
Within Session, a region is a "pointer" to a particular *audio file* or portion of an audio file. Regions can be dragged from the *Audio Regions list* into a *track*. As the program plays back, a region instructs Session to play its associated audio — either an entire audio file, or, if any editing has been performed, a user-defined portion of an audio file.

Regions List
A holding area for any *regions* that you create. Regions can be dragged from the Audio Regions List into a track and arranged as desired.

Resize
See *Trim*.

Reverberation; Reverb
A type of processing that many *digital effects processors* and some *DSP* programs can create. Reverb adds a sense of ambience to recordings, simulating how sound reflects around and reverberates within concert halls, cathedrals, arenas, and even such pedestrian spaces as a shower stall or coffin. Technically speaking, reverb is similar to *delay*, but the sound is generally diffuse, so that no individual echoes or other reflections are distinct. Reverb processing simulates "early reflections" — those reflections of sound that would be amongst the first to reach our ears in a concert hall or other space — as well as later reflections. Natural reverb settings have a decay, or duration, of up to a second or two.

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In practice, judicious use of reverb is an excellent way to add a professional "sheen" to a production, but too much reverb — or too long a decay setting — can "muddy" a production's sound, making everything indistinct and lost in a "wash" of reverb.

Sample Rate

Sample rate is one of two main specifications that describe digital audio quality (the other is bit-rate resolution). In digital audio, sample rate refers to how frequently incoming audio is "sampled" as its converted from an analog to a digital signal. A sample is like an instantaneous "snapshot" of sound; if you take lots of snapshots and string them together, you can get an accurate representation of what's going on, sound wise — just like a movie is a strung-together collection of pictures. High-quality audio is usually sampled 44,100 times, or cycles, per second, and is often referred to as a "44.1 kiloHertz" (44.1kHz) sample rate. Some people use a higher sample rate of 48kHz, but this has to be "sample rate converted" if it's to be used on an audio CD, so it's generally preferable to stick to 44.1kHz. Lower sample rates are also possible: Broadcasters sometimes work at a 32kHz rate, and many games and other multimedia programs use audio sampled at a 22kHz or even 11kHz rate.

Why all these different rates? Sample rate has a direct bearing on two things: audio quality, and file size. In terms of quality, the sample rate is what defines the upper frequency response of audio; for reasons that are a bit too technical for the scope of this glossary, the upper frequency response is roughly half the sample rate. For instance, 44.1kHz sampled audio has a frequency range of up to 22kHz — somewhat beyond most people's hearing range. Audio sampled at an 11kHz rate, however, has a frequency range of up to 5.5kHz — which, consequently, will sound a bit "muffled" to almost everyone's ears. (The typical telephone connection is good to about 6kHz.) On the other hand, a one-minute 16-bit 44.1kHz-sampled mono file takes up roughly 5MB of hard disk space, whereas a one-minute 16-bit 22kHz file will use roughly half as much space. As usual in life, it's a trade-off between quality and quantity. Also see *Bit-Rate Resolution.*

SCSI

Small Computer System Interface, a standard means of connecting *hard disks*, scanners, and other important peripherals to personal computers, especially Macintosh computers. The basic SCSI specification allows up to seven devices to be connected to a SCSI "chain," usually via 25-pin and 50-pin SCSI connectors. The last device physically connected to the SCSI chain needs to be "terminated," either through an electronic switch, or more commonly, with a *terminator block* that is attached to one of the device's SCSI connectors.

Seek Time

Refers to the average time it takes the heads of a *hard disk* to locate a desired portion of data. Digital audio applications — such as Session — require seek times in the neighborhood of 18 milliseconds or faster, depending upon how many tracks are being played back at once.

Selector

A tool within Session. The Selector lets you select *regions* or portions of regions with an "I-beam"-like tool, in order, for instance, to *cut*, *copy*, or *paste*.

Sequencer

A *MIDI*-based computer program (or stand-alone device) that records MIDI data such as notes, duration, and so forth. A MIDI sequencer is essentially a *multitrack recording* system for synthesizers, drum machines, and other MIDI-capable devices. Using *OMS*, Session can communicate with, and work in tandem with, a MIDI sequencer.

Session

1) Digidesign's audio production program; 2) a working session file, much like a word processing document or a graphics program file. Each separate project you work on should be stored as its own session.

Shielded Monitor Speakers

Most loudspeaker designs utilize large magnets, which — when placed near a computer video display — can bend and otherwise distort onscreen images. Many professional *monitor speakers*, consequently, are shielded with an internal antimagnetic barrier or foil, which allows them to be placed directly adjacent to a video monitor with no visual distortion.

Signal Levels

There are a wide range of different audio signals, from the relatively faint level that comes directly from a microphone to the sometimes dangerously high-voltage signal that travels out of a power amp on its way to a loudspeaker. (While digital audio signals travel through wiring at measurable levels, the levels we'll discuss here — and the only ones you need to worry about — are all analog levels, because digital levels are generally standardized.)

There are also a wide range of ways to measure signal level, though most are some variation of the *decibel* (dB), such as dBV (typically used for home stereo and other "semi-pro," unbalanced gear; dBm (for professional, balanced gear); and dBA (for measurement of acoustic sounds, like that jackhammer outside your window that you'd like to report to the proper authorities). The actual differences between these different levels are more complicated than anyone other than a professional engineer needs to understand, but fortunately, there is a common scale that we can use — the "dBu" (also known as dBv, which is not to be confused with dBV; we told you this could get complicated).

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In a nutshell, microphone signals are at the bottom of the scale, pumping out a puny -60 to -50dBu (-50dBu is a higher signal than -60dBu). The next step is "instrument level," such as the level produced by an electric guitar; this ranges in the -50dBu to -20dBu range. Now we some to "line level" signals, which refer to the most common connections between components. Most home stereos, semi-pro recording systems, computer sound cards, and other such unbalanced products operate at a "nominal" line level of -10dBu. (We say nominal because the actual levels varies as output levels vary; -10dBu represents the theoretical maximum line level, although almost every -10dBu product can produce or receive levels as high as -5dBu, 0dBu, or higher; see *headroom*.)

Next stop on our way up the chain: +4dBu, which is the nominal line level for most professional balanced gear. As long as we're using the dBu scale, a +4 product's line level can be said to be 14 decibels "hotter" (stronger) than a -10dBu product, which — all other things being equal — translates to a somewhat better *signal-to-noise (S/N) ratio*. Finally, some broadcasters set their gear to operate at +8dBu. Most electronic keyboards, drum machines, and other MIDI gear, by the way, have variable outputs that go up to -10dBu, 0dBu, or even +4dBu.

It's always important to match output signals to the proper type of input, to avoid a level mismatch. For instance, if you plug the +4dBu output of a pro-quality effects box into a microphone input, you'll almost certainly overload the input, and cause distortion. If you connect a low level signal — like the output of an electric guitar — into a -10dBu line level input, you'll probably hear something, but it will be so faint that you'll need to crank up the input, which means you'll hear an excess of noise. This is why microphones should always be connected to a microphone-level input, and why line level-devices should always be connected to the proper line-level input. If in doubt, check with your dealer or the product manufacturer.

Mixing consoles, digital audio workstations, and other audio equipment typically use *VU meters* to measure relative differences in level.

Signal Processor
An "outboard" analog or digital box — or type of software — that processes sound in one or more specialized ways, such as an *equalizer*, a *delay*, and so forth.

Signal-To-Noise Ratio (S/N Ratio)
Similar to *dynamic range*. Whereas dynamic range describes the span between a signal's lowest and highest possible levels, however, the S/N ratio describes the span between a signal's highest (or near-highest) level and the average "residual" noise of the device which is being measured. Effectively, the S/N ratio describes how "quiet" a signal is — how free it is of hiss, hum, and other relatively continuous noise. The S/N ratio is expressed in decibels; the higher the number, the better. Professional digital products usually exhibit a S/N ratio of 85dB to 93dB; analog cassette recordings, on the other hand, might exhibit a S/N ratio of 45dB to 75dB, depending upon the type of noise reduction is used (if any).

Sometimes, you'll find specifications that are expressed simply as signal-to-noise (without the ratio part). In this case, the figures are shown in "–" decibels, which reflect how many decibels lower the noise is compared to the signal. In this case, for instance, we would say that professional digital products usually exhibit a S/N specification of -85dB to -93dB.

SMPTE/EBU Timecode
The standard timecode of Society of Motion Picture and Television Engineers/European Broadcasters' Union. Divides time into hours:minutes:seconds:frame:subframes. Typical frame-per-second (fps) rates include 24 (film), 25 (European video), 29.97 (North American Color Video), and 30 (North American B&W video). 30fps timecode can be in *drop-frame* or *non-drop-frame* formats. Also see *MTC*.

SND
See *Sound Resource File Format*.

Solo
A common *mixing console* feature that is used to isolate one or more *channels* within the mix, so that they can be monitored alone — without hearing any tracks that have not been soloed. Solo is rather like the opposite of *mute*. Session includes a solo function for each of its tracks.

Sound Card
See *audio card*.

Sound Designer II (SDII) File Format.
One of the two most common high-quality digital audio file formats for the Macintosh. Created by Digidesign, the Sound Designer II format is used primarily for "CD quality audio" — 16-bit audio at a 44.1kHz sampling rate, although some users work at a 48kHz sampling rate. Named for Digidesign's Sound Designer II professional audio editing program.

Sound Resource (SND) File Format
Apple's sound resource file type, used primarily by the Macintosh System Software, and by some Macintosh software applications. Session does not directly support this type of file in its sessions. If you wish to use an SND file within Session, first import it using the Import Audio File command. The SND format is best if you plan to use your audio in applications or situations that do not support the Sound Designer II format, such as a System alert sound (beep).

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Sources & Destinations
In audio terms, signals are routed from sources (such as a microphone) to destinations (such as a microphone input on a mixer).

S/PDIF Digital Interface
Describes an industry standard for transferring digital audio between audio devices; essentially a "semi-pro" version of the *AES/EBU* standard. Stands for "Sony/Philips Digital Interface Format." Most S/PDIF connections utilize an *RCA* connector, which is able to carry two channels of digital audio (as opposed to one channel of analog audio). Some *audio cards*, including Digidesign's Audiomedia II, is equipped with S/PDIF *I/Os*.

Spotting
An *audio post-production* process of aligning audio events to visual events. Session has a Grid mode that allows you to spot regions to particular time code events.

Stereo
Audio comprised of two discrete channels — a left, and a right. "Stereophonic" hearing allows us to locate sounds within the *stereo field*. Also see *pan control*.

Stereo Field
An imaginary "sound stage" that defines where the *stereo* panorama where a particular sound is placed — from left to center to right. Also see *pan control*.

Submix; Subgroup
See *Grouping*.

Superdirectional Microphone
See *Hypercardioid Microphone*.

Synchronization
The process of having two or more audio, video, or other devices run together at exactly the same speed (usually during playback, though sync during record, fast-wind, and "crawls" is possible). Professional applications require the use of an external or plug-in "transport synchronizer." With an affordable DAW, such as Digidesign's Session, synchronization is usually performed either between Session and a sequencer using *OMS*, or in *Trigger Sync* mode, where an external *SMPTE*-to-*MTC* converter (such as Opcode's Studio 4 Interface) sends a "start" signal to the DAW at a predetermined start time.

Sync Point
A user-defined point within a *region* which defines where and how that region will be placed, for instance, when moved within a grid, or when *spotted* to a particular time.

Take
[noun] Originally a term from film production (as in "Scene 7, Take 3"), recording engineers have expropriated it to refer to each attempt, however successful, at recording a desired section of audio or music. For instance, a musician attempting to recording an important musical solo, or an actor attempting a *voice-over*, frequently will require several takes before successfully "nailing" the right one. Through careful editing, a final track can be assembled from a variety of takes, using portions of a variety of takes. With Session's *nondestructive recording* capabilities, multiple takes can be recorded on each track.

Terminator Block
A small accessory that is usually attached to the last device in a *SCSI* chain.

THD
Total Harmonic Distortion; see *Harmonic Distortion*.

Thermal Recalibration
Some *hard disks*, particularly older ones, perform a function called thermal recalibration, which can interrupt the playback of digital audio or video files. This happens when the hard disk's internal platter(s) expand slightly from the heat generated by continual use, which in turn causes the hard disk pause momentarily to recalibrate the relationship of the head to the platter. Hard disks that are certified "A/V compatible" are free of thermal recalibration, and are the best choice for use with any *DAW*, including Session, if interruption-free audio playback is desired.

Track
Refers originally to a continuous longitudinal (lengthwise) strip of a magnetic tape that would hold an audio recording. Tape designed for stereo recording has two tracks (or four tracks, if it's designed to be flipped over to record on a second "side", such as with an analog audio cassette). In recent years, a track has expanded beyond tape, and beyond this literal definition. A typical MIDI sequencer, for instance, can record dozens of MIDI tracks, usually overdubs of one or more MIDI instruments, with each track on a different *MIDI channel*. A DAW, on the other hand, such as Session, can also record overdubs of different audio or musical events on multiple different tracks. (Session's number of available tracks varies depending upon the type of computer and audio card being used.) Also see *Multitrack Recording*.

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Transport Window
Within Digidesign's Session, the *Transport* window contains controls for (in order) Return to Zero, Rewind, Stop, Play, Fast Forward, Record, and Loop Playback. The Transport also contains controls for creating *Markers*, which allow you to quickly navigate to a desired point in a recording. The Marker controls can be displayed or hidden by clicking the *grow box* at the upper right corner of the Transport.

TRS (Tip-Ring-Sleeve) Connector
Most commonly, a *balanced* version of a 1/4" connector, in which the "tip" of the plug is "hot," the "ring" is "cold," and the "sleeve" is the ground. TRS connectors can also be used as stereo connectors, where the tip is usually "right channel," the ring is "left," and the sleeve is "ground." The standard headphone plug, as found with home stereo headphones and receivers, is a 1/4" TRS stereo plug. (The connector found on most portable stereos is typically an 1/8" TRS stereo plug.)

Unbalanced Audio
See *Balanced & Unbalanced Audio.*

Unidirectional Microphone
See *Cardioid Microphone.*

Virtual Track
An onscreen track that isn't assigned a *voice*. See *Multitrack Recording.*

Voices
With a *DAW*, this refers to the number of tracks that can be played back simultaneously. See *Multitrack Recording.*

VTR
A video tape recorder. Professionals use this term instead of "VCR" (for video cassette recorder) since some professional VTRs use open-reel tape instead of cassette tape.

VU; VU Meter
Stands for Volume Unit. Recording equipment and programs such as Session use VU meters to indicate relative signal levels. Each device's VU meters are usually calibrated so that "0VU" represents indicates a signal with the same level as the device's nominal operating level.

For instance, when a –10dBu device (such as Session, used with Digidesign's Audiomedia card, or a home stereo analog cassette recorder) receives a signal of –10dBu strength, its input meters will indicate a level of 0VU. If the same device receives a signal of –25dBu strength, its input meters will indicate –15dBu. Alternately, when a professional-style +4dBu device (such as Pro Tools, with the 882 *audio interface*) receives a signal of +4dBu strength, its input meters will indicate 0VU. If the same device were to receive a signal of +7dBu strength, its input meters will indicate +3VU.

VU meters are usually calibrated from a low of –40VU or –20VU to a high of about +6VU or +10VU.

.WAV
Pronounced "wave"; Microsoft's Audio File Format for Windows. Many Windows software applications, and some Mac applications, support this file format. Digidesign's Session does not directly support this type of file in its sessions. If you wish to use a .WAV file within Session, you will first have to import it using the Import Audio File command. The .WAV format is best if you plan to use bounced audio in an application that supports or requires .WAV format files.

Waveform
A means of visually representing a sound. When sound *regions* are imported into Session's *Edit window*, they can be viewed in waveform mode, which displays the changing amplitude of the region over time. Waveform display is very useful for editing sounds; an unwanted click, for instance, may appear as a sharp "spike" in the waveform, allowing you to *select* it and *cut* it.

XLR (3-pin) Connector
A common connector for professional audio devices, that utilizes *balanced audio* connections. "Male"-style XLR connectors are used as audio output connectors, and have three pins. In most professional audio gear, pin 1 is universally the ground conductor; pin 2 is the "hot" conductor; and pin 3 the "cold" conductor. (See *Balanced & Unbalanced Audio.*) Some devices, however, reverse the roles of pins 2 and 3. "Female"-style XLR connectors are used as audio inputs, and have three holes to accommodate the male connector's pins. Also called "Canon" connector, for the brand name owned by ITT Corporation.

Z
Another term for *impedance.*

Zoom
A function within Digidesign's Session program, that allows you to view waveform displays within the *Edit window* with greater detail.

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GLOSSARY

A-B Roll: Videotape editing arrangement where scenes on tape are played alternately on VTRs A and B and recorded on VTR C. Typically, the final output recorded on VTR C contains some scenes from VTR A and some scenes from VTR B with transitions (cuts, mixes, wipes, etc.) between the scenes.

Aberrations: Certain aberrations degrade the image formed by a lens.

Absorption Loss: In telecommunications, attenuation of the optical signal within the fiber optic transmission medium. Usually specified in terms of dB/km.

AC/DC Coupling: May also be called simply DC coupling. Coupling between circuits which accommodates the passing of both AC and DC signals.

ADC: Abbreviation of analog-to-digital converter.

Address: 1. A precise frame location on a videotape, usually identified by a time code number. 2. A memory location or device identifier in microprocessor and computer terminology.

Aliasing: Undesirable "beating" effects caused by sampling frequencies being too low to faithfully reproduce image detail. Examples are:

i). Temporal aliasing—e.g. wagon wheel spokes apparently reversing, also movement judder seen in standards converters with insufficient temporal filtering.

ii). Raster scan aliasing—twinkling effects on sharp horizontal lines.

Raster scan aliasing and its horizontal equivalent are often seen in older digital effects devices as detailed images are compressed, due to insufficient filtering. Aliasing is also often used to describe the unpleasant stepped images if unfiltered angled lines are presented upon the raster lines of a TV system.

Amplitude: The magnitude of a signal in voltage or current. Frequently expressed in terms of peak, peak-to-peak or RMS.

Analog: The characteristic of varying continuously along a scale as opposed to increasing or decreasing in fixed steps. Voltage, pressure, speed, etc. are often measured in analog terms. A continuously variable system or device. Continuous tone film and a volume control on an average radio or record player are analog.

Analog-to-Digital Converter: (ADC, A/D, A-to-D) A circuit that uses digital sampling to convert an analog signal into a digital representation of that signal.

Anti-Aliasing: A procedure employed to eliminate or reduce, by smoothing and filtering, the aliasing effects. Aliasing, which is a disturbing effect created on a video image where vertical lines are either too close together or where a lot of high frequency information is concentrated in a limited area of the screen (like from computer generated text and graphics) creates "beating," "crawling" and strobing interference.

Aperture: The opening of a lens which controls the amount of light reaching the surface of the pickup device. The size of the aperture is controlled by the iris adjustment. By increasing the f-stop number (f/1.4, f/1.8, f/2.8, etc.) less light is permitted to pass to the pickup device.

APL (Average Picture Level): The average level of the active video (portion of video between blanking pulses), expressed as a percentage or in IRE.

Archive: Long-term off-line storage. In digital systems, pictures are generally archived onto some form of hard disc, ½" magnetic tape, floppy disc or 8mm cartridge.

Artifacts: Undesirable elements or defects in a video picture. These may occur naturally in the video process and must be eliminated in order to achieve a high quality picture. Most common are cross color and cross luminance.

Aspect Ratio: 1. The ratio of television picture width to height. In NTSC and PAL video, the present standard is 4:3. 2. The ratio of wipe pattern width to height.

Assemble Edit (Assemble Mode): An editing mode that replaces all signals on the record tape (video, audio, control and time code tracks) with new signals. See also Insert Edit.

Astigmatism: The uneven foreground and background blur that is in the image.

Asynchronous: Lacking synchronization. In video, a signal is asynchronous when its timing differs from that of the system reference signal. A foreign video signal is asynchronous before it is treated by a local frame synchronizer.

Attenuator: A circuit that provides reduction of the amplitude of an electrical signal without introducing appreciable phase or frequency distortion.

Audio Bridge: In telecommunications, a device that mixes multiple audio inputs and feeds back composite audio to each station, minus that station's input. Also known as a mix-minus audio system.

Audio-Follow-Video (AFV): An operational mode in which audio and video switchers are tied together so when the operator selects the video source, the audio simultaneously and automatically switches to the same source.

Auto Assembly: In video editing, the editing system automatically records all edits listed in the edit decision list. This allows the operator to complete an edit decision list and then let the editing system perform all of the recording automatically.

Axis: Relating to digital picture manipulation, the X axis is a horizontal line across the center of the screen, the Y axis is a vertical line and the Z axis is in the third dimension, perpendicular to the X and Y axes and indicates depth and distance.

Background Video: 1. Video that forms a background scene into which a key may be inserted. 2. A solid color video output generated by the background generator within a device, such as a production switcher, for use as background video in key effects.

Back Light: A fixture that is often not properly applied or overlooked completely. The main function of the back light is to separate the individual subjects from the background and give them depth and dimension.

Back Porch: The portion of a video signal that occurs during blanking from the end of horizontal sync to the beginning of active video. The blanking signal portion which lies between the trailing edge of a horizontal sync pulse and the trailing edge of the corresponding blanking pulse. Color burst is located on the back porch.

Bandwidth: The complete range of frequencies over which a circuit or electronic system can function with minimal signal loss, typically less than 3dB. The information carrying capability of a particular television channel. In PAL systems the bandwidth limits the maximum visible frequency to 5.5MHz, in NTSC, 4.2MHz. The CCIR 601 luminance channel sampling frequency of 13.5MHz was chosen to permit faithful digital representation of the PAL and NTSC luminance bandwidths without aliasing.

Baseband: The frequency band occupied by the aggregate of the signals used to modulate a carrier before they combine with the carrier in the modulation process.

Base and Fill Lights: Base and fill lights, commonly referred to as "scoops," provide a soft-edged field of light which is used to provide basic illumination of the subject, to fill in the areas not highlighted by the key light, to illuminate the background and to soften shadows caused by key lights.

Baud: A unit of signalling speed equal to the number of signal events per second. Baud is equivalent to bits per second in cases where each signal event represents exactly 1 bit. Often the term baud rate is used informally to mean baud, referring to the specified maximum rate of data transmission along an interconnection. Typically, the baud settings of 2 devices must match if the devices are to communicate with one another.

Bearding: Video distortion that appears as short black lines extending to the right of bright objects within a scene.

Betacam, Betacam Format: Portable camera/recorder system and related equipment originally developed by Sony; the name may also be used for just the recorder or for the interconnect format; Betacam uses a version of the (Y, R-Y, B-Y) component set.

Betacam SP: A Superior Performance version of Betacam. SP uses metal particle tape and a wider bandwidth recording system. The interconnect standards are the same as Betacam. There is also limited tape interchangeability with standard Betacam.

Bias: Current or voltage applied to a circuit to set a reference operating level for proper circuit performance, such as the high frequency bias current applied to an audio recording head to improve linear performance and reduce distortion.

Binary: A base-2 numbering system using the 2 digits 0 and 1 (as opposed to 10 digits (0-9) in the decimal system). In computer systems, the binary digits are represented by 2 different voltages or currents, 1 corresponding to 0 and another corresponding to 1. All computer programs are executed in binary form.

Bipolar: A signal that contains both positive-going and negative-going amplitude. May also contain a 0 amplitude state.

Bit (Binary Digit): The smallest part of information in a binary notation system. A bit is a single 1 or 0. A group of bits, such as 8 bits or 16 bits, compose a byte. The number of bits in a byte depends upon the processing system being used. Typical byte sizes are 8, 16 and 32.

Bitmap: A pixel-by-pixel description of an image. Each pixel is a separate element. Also referred to as a raster image.

Black also **Color Black, Blackburst:** 1. A composite color video signal. This signal has composite sync, reference burst and a black video signal which is usually at a level of 7.5 IRE (0.05V) above the blanking level. 2. Fade-to-black between scenes.

Blanking (BLKG): 1. The time period when picture information is shut off. Blanking is a voltage level which is at or below black picture level and acts as a signal to turn off the scanning beam. Synchronizing pulses which control invisible retrace of scanning are active during the blanking period. 2. A standard signal from a sync generator used to create blanking in video.

Blanking Level: Also known as pedestal, the level of a video signal which separates the range that contains the picture information from the range that contains the synchronizing information. The level of the front and back porches. Zero IEEE units.

Blooming: 1. The defocusing of regions of the picture where brightness is excessive. 2. On video monitors, adjusting the white levels so they are just at the point of leaving gray and becoming white.

BNC: Abbreviation of bayonet Neill-Concelman. A cable connector used extensively in television and named for its inventor.

Bridge: 1. A type of network circuit used to match circuits to each other, ensuring minimum transmission impairment. 2. To place one circuit parallel to another.

Brightness: In NTSC and PAL video signals, the brightness information at any particular instant in a picture is conveyed by the corresponding instantaneous DC level of active video. Brightness control is an adjustment of setup (black level, black reference).

Broadband: 1. Having an essentially uniform response over a wide range of frequencies. 2. Capable of handling frequencies greater than those required for high grade voice communications (higher than 3 to 4kHz).

Buffer: 1. A circuit or component which isolates one electrical circuit from another. 2. A digital storage device used to compensate for a difference in the rate of flow of information or the time of occurrence of events when transmitting information from one device to another. 3. In telecommunications, a protective material used in cabling optical fiber to cover and protect the fiber. The buffer material has no optical function.

Burst (Color Burst): 7 to 9 cycles (NTSC) or 10 cycles (PAL) of subcarrier, placed near the end of horizontal blanking to serve as the phase (color) reference for the modulated color subcarrier. Burst serves as the reference for establishing the picture color.

Burst Flag (BF): A pulse used to gate the color reference subcarrier (burst) onto the back porch of each horizontal blanking interval. Also called burst gate (BG).

Burst Vector: In composite video signals, the amplitude and angle of the color reference signal.

Byte: Unit of memory in a computer. Consists of 8 bits. Generally, one byte expresses image intensity at one point (pixel) of an image in one channel. Or it can represent one letter, number or symbol in the ASCII code.

Cable Equalization: The process of altering the frequency response of a video amplifier to compensate for high frequency losses in coaxial cable.

Camera Control Unit (CCU): A separate electronics frame connected to a video camera head to supply it with power and control. The CCU also provides encoding and/or processing of the video signal. Operator controls available at the CCU usually include video levels, color balancing and iris control.

Candlepower: The unit measure of an incident light.

Capacitor: A device that stores electrical energy. It allows the apparent flow alternating current while blocking the flow of direct current. The degree to which it allows AC current flow depends on the frequency of the signal and the size of the capacitor. Capacitors are used in filters, delay line components, couplers, frequency selectors, timing elements, voltage transient suppression, etc.

Cardioid: A type of microphone with sound pickup characteristics resembling a heart-shape sphere. The cardioid microphone is used in specific applications where a pickup characteristic of this kind is needed.

Carrier Wave: A single frequency wave which, when transmitted, is modulated by another wave containing information.

CAV (Component Analog Video): A video format in which 3 separate video signals represent luminance and color information. Each signal consists of an analog voltage that varies with picture content. Also called analog component.

CCD: Abbreviation of Charge Coupled Device. A device that stores samples of analog signals. Used in cameras and telecines as an optical scanning mechanism. Advantages include good sensitivity in low light and absence of burn-in and phosphor lag found in CRTs.

CCD Array: A device that mounts many CCDs together to allow for capture of many pixels at the same time. Currently, 4 megapixels CCD arrays are in production. This means that 4,194,304 pixels of light can be converted into digital values at the same instant in such an array.

CCD Color Scanner: An input scanner using a lens and a linear CCD array to produce the scan raster. The array (one pixel wide by several thousand long) is "stepped" sideways across the focal point of the lens, each step producing one complete scan line signal.

CCIR 601: An international standard for component digital television that was derived from the SMPTE RP125 and EBU 3246E standards. CCIR 601 defines the sampling systems, matrix values and filter characteristics for both Y, Cr, Cb and RGB component digital television. It establishes a 4:2:2 sampling scheme at 13.5MHz for the luminance channel and 6.75MHz for the chrominance channels with 8-bit digitizing for each channel. These sample frequencies were chosen because they work for both 525-line 60Hz and 625-line 50Hz component video systems. The term 4:2:2 refers to the ratio of the number of luminance channel samples to the number of chrominance channel samples; for every 4 luminance samples, the chrominance channels are each sampled twice. The D1 digital videotape format conforms to CCIR 601.

CCIR 656: The international standard defining the electrical and mechanical interfaces for digital television equipment operating according to the CCIR 601 standard. CCIR 656 defines both the parallel and serial connector pinouts as well as the blanking, sync and multiplexing schemes used in both parallel and serial interfaces.

CGA: Color Graphics Adaptor.

Character Generator: Reproduces recognized font styles from a computer type keyboard—usually provides multiple screen storage and is capable of background colorization from video display.

Checkerboard Assembly: In video editing, a nonsequential method of auto assembly. The computerized editing system records and edits from the videotape playback reels currently in use, leaving gaps that will be filled later by subsequent reels. Also called B-mode assembly. See Auto Assembly.

Chroma Crawl: An artifact of encoded video also known as dot crawl or cross luminance. Occurs in the video picture around the edges of highly saturated colors as a continuous series of crawling dots and is a result of color information being confused as luminance information by the decoder circuits.

Chroma Gain (Chroma, Color, Saturation): In video, the gain of an amplifier as it pertains to the intensity of colors in the active picture.

Chroma Key (Color Key): A video key effect in which one video signal is inserted in place of areas of a particular color in another video signal. For example, a weatherman stands in front of a blue wall with a camera focused on him. The camera signal feeds a chroma keyer which detects the blue in the blue wall and replaces it with video from another camera, such as video of a weather map. Thus, the finished key makes the weatherman appear to be standing in front of the weather map.

Chromaticity: The attribute of light combining hue and saturation, independent of intensity. The color perceived is determined by the relative proportions of the 3 primary colors. The color quality of light, defined by wavelength and purity.

Chrominance: The color part of a signal, relating to the hue and saturation but not to the brightness or luminance of the signal, e.g. black, gray and white have no chrominance, but any colored signal has both chrominance and luminance. U,V: Cr,Cb: I,Q: (R-Y), B-Y) represent the chrominance information of a signal. See: YUV and YIQ.

Chrominance-to-Luminance Intermodulation (Crosstalk, Cross-Modulation): An undesirable change in luminance amplitude caused by superimposition of some chrominance information on the luminance signal. Appears in a TV picture as unwarranted brightness variations caused by changes in color saturation levels.

Clamp, Clamping: The circuit or process that restores the DC component of a signal. A video clamp circuit, usually triggered by horizontal synchronizing pulses, re-establishes a fixed DC reference level for the video signal. Some clamp circuits clamp sync tip to a fixed level, and others clamp back porch (blanking) to a fixed level. A major benefit of a clamp is the removal of low-frequency interference, especially power line hum.

Clip: 1. In keying, the trigger point or range of a key source signal at which the key or insert takes place. 2. The control which sets this action. To produce a key signal from a video signal, a clip control on the keyer control panel is used to set a threshold level to which the video signal is compared. 3. In digital picture manipulators, a menu selection that blanks portions of a manipulated image that leave one side of the screen and "wrap" around to enter the other side of the screen.

Clipping Level: An electronic limit to avoid overdriving the audio or video portion of the television signal.

C-Mount: A C-Mount is generally the standard mounting means for attaching a lens to a camera. Normally, a C-Mount uses a 1"-32 thread. With a C-Mount, the dimension from the banking shoulder of the lens mounting thread to the image plane of the camera is 0.690" regardless of the kind of lens used.

CMYK: A color encoding system used by printers in which colors are expressed by the "subtractive primaries" (cyan, magenta and yellow) plus black (called "K" or keyline since black, keylined text appears on this layer). The black layer is added to give increased contrast and range on printing presses. See RGB.

Color Difference Format: A video signal set that includes color difference signals. Betacam and MII, for example, are names of 2 widely used color difference formats.

Color Difference Signal: A video color signal made by subtracting luminance and/or color information from one of the primary color signals (red, green or blue). In the Betacam color difference format, for example, the luminance (Y) and color difference components (R-Y and B-Y) are derived as follows:

Y = 0.3 Red + 0.59 Green + 0.11 Blue
R-Y = 0.7 Red - 0.59 Green - 0.11 Blue
B-Y = 0.89 Blue - 0.59 Green -0.3 Red

The G-Y color difference signal is not created because it can be reconstructed from the other 3 signals. Other color difference conventions include SMPTE, EBU-N10 and MII. Color difference signals should not be referred to as component video signals. That term is reserved for the RGB color components. In informal usage, the term component video is often used to mean color difference signals.

Color Field: In the NTSC system, the color sub-carrier is phase-locked to the line sync so that on each consecutive line, subcarrier phase is changed 180° with respect to the sync pulses. In the PAL system, color subcarrier phase moves 90° every frame. In NTSC this creates 4 different field types; in PAL there are 8. In order to make clean edits, alignment of color field sequences from different sources is crucial.

Color Frame: In color television 4 (NTSC) or 8 (PAL) properly sequenced color fields compose one color frame.

Color Phase: The timing relationship in a video signal which is measured in degrees and keeps the hue of a color signal correct. Color information is encoded in the video signal as the difference in phase between the sine peaks of the chrominance signal and the color burst subcarrier signal. If the 2 signals overlap exactly, then the phase difference is designated as 0°. If the sine signals do not overlap, the color phase can vary from 0° to 360°. Each shift in the color phase represents a specific tint on the screen. If 2 sine waves are shifted one from the other by 180°, then the colors are totally inverted.

Color Subcarrier: The 3.58MHz signal which carries color information. This signal is superimposed on the luminance level. Amplitude of the color subcarrier represents saturation and phase angle represents hue.

Color Temperature: Indicates the hue of the color. It is derived from photography where the spectrum of colors is based upon a comparison with the hues produced when a special metal body is heated from red through yellow to blue, which is the hottest. Color temperature measurements are expressed in Kelvin.

Comb Filter: An electrical filter circuit that passes a series of frequencies and rejects the frequencies in between, producing a frequency response similar to the teeth of a comb. Used on encoded video to select the chrominance signal and reject the luminance signal, thereby reducing cross chrominance artifacts, or conversely, to select the luminance signal and reject the chrominance signal, thereby reducing cross luminance artifacts. Comb filtering successfully reduces artifacts but may also cause a certain amount of resolution loss in the picture.

Component: The normal interpretation of a component video signal is one in which the luminance and chrominance are sent as separate components, e.g. analog components in MII and Betacam VTRs, digital components YCRCB in CCIR rec 601. RGB is, however, also a component signal. Component video signals retain maximum bandwidth, unlike composite systems.

Composite: A composite video signal is one in which the luminance and chrominance information have been combined using one of the coding standards: NTSC, PAL, SECAM, etc.

Composite Sync: A signal consisting of horizontal sync pulses, vertical sync pulses and equalizing pulses only, with a no-signal reference level.

Composite Video: A mixed signal comprised of the luminance (black and white), chrominance (color), blanking pulses, sync pulses and color burst.

Contrast: The range of light and dark values in a picture or the ratio between the maximum and the minimum brightness values. Low contrast is shown mainly as shades of gray, while high contrast is shown as blacks and whites with very little gray. It is also a TV monitor adjustment which increases or decreases the level of contrast of a televised picture.

Control Track Frame Pulse: A pulse laid down on videotape by a videotape recorder to identify the frame locations on the videotape. This enables the VTR to lock up correctly framed during playback.

Cross Color: This defect manifests itself as spurious rainbow patterns on highly textured objects like a striped shirt or tweed jacket. Cross color defect is attributed to the make-up of the NTSC signal which mixes the high luminance and chrominance information in the same composite baseband spectrum.

Crosstalk: 1. Undesired transmission of signals from one circuit into another circuit in the same system. Usually caused by unintentional capacitive (AC) coupling. 2. Signal interference from one part of a videotape to another.

CT/Continuous Tone: A picture file. CT is an abbreviation of continuous tone (also called contone). CT files are created by either scanning a picture into the system or by generating a CT image internally. Each pixel in a CT file uses one byte each for its red, green and blue values, allowing up to 256 density levels per color and more than 16 million different mixture colors.

Cyclorama Lights: Cyclorama lights are designed to create a smooth lighting effect on a backdrop or cyclorama.

D1: A component digital videotape recording format that conforms to the specifications set in the CCIR 601 standard.

D2: An 8-bit composite digital videotape recording format in which the composite video signal is digitized by sampling it at the rate of 4 times the frequency of subcarrier ($4f_{sc}$). The $4f_{sc}$ frequency in NTSC is 14.3MHz and 17.7MHz in PAL.

D3: An unofficial term for a composite digital videotape recording format invented by Panasonic.

D5: A component digital videotape recording format that conforms to the specifications set in the CCIR 601 standard; Panasonic format.

D-to-A Converter: DAC—A device used to convert digital signals to analog signals.

DAT: Digital Audio Tape. A system developed initially for recording and playback of digitized audio signals, maintaining signal quality equal to that of a CD. Recent developments in hardware and software might lead to a similar inexpensive system for video recording and playback.

Data Compression: A technique that provides for the transmission or storage, without noticeable information loss, of fewer data bits than were originally used when the data was created.

dB (decibel): A measure of voltage, current or power gain equal to 1/10 of a Bel. Given by the equations 20 log Vout/Vin, 20 log Iout/In, or 10 log Pout/Pin.

Decoder: A device used to recover the component signals from a composite (encoded) source. Decoders are used in displays and in various processing hardware where component signals are required from a composite source, i.e., composite chroma keying of color correction equipment, etc.

Degauss: To demagnetize recording and playback heads, tape.

Delay Line: An artificial or real transmission line or equivalent device designed to delay a wave or signal for a specific length of time.

Demagnetize: To remove magnetism; to erase magnetic tape.

Demodulator: TV demodulators strip the video and audio signals from the carrier frequency. The composite video and audio can then be used as any other video or audio feed for studio use.

Depth of Field: The front to back zone in a field of view which is in focus in the televised scene. With a greater depth of field, more of the scene (near to far), is in focus.

Deserializer: A device that converts parallel digital information to serial.

Differential Gain: A change in subcarrier amplitude of a video signal caused by a change in luminance level of the signal. The resulting TV picture will show a change in color saturation caused by a simultaneous change in picture brightness.

Differential Phase: A change in subcarrier phase of a video signal caused by a change in luminance level of the signal. The hue of colors in a scene change with the brightness of the scene.

Digital: Circuitry in which data carrying signals are restricted to either of 2 voltage levels, corresponding to logic 1 or 0. A circuit which has 2 stable states: high or low, on or off.

Digital Components: Component signals in which the values for each pixel are represented by a set of numbers.

Digital Disc Recorder: A system mainly intended for post-production purposes, allowing a person to record short scenes on a digital disc. The advantages of this system for editing purposes are extremely fast access to any point on the disc, elimination of dropout and very fast shuttle speed back and forth. Several digital formats of data storage exist, developed specifically for the disc recorder by the manufacturer, without a universal standard.

Digitizing Pad: A device that translates drawings from a tablet and stylus to a digital video format.

Disc: A flat circular plate, coated with a magnetic material, on which data may be stored by selective magnetization of portions of the surface. May be a flexible, "floppy" disc or rigid "hard" disc.

Dispersion: The characteristic of a light-conducting medium that causes the medium to transmit light of different frequencies at different velocities. Dispersion causes the refractive index of a given medium to vary as a function of wavelength. As it relates to optical fiber, this property influences both the effective numerical aperture and the bandwidth of an optical fiber.

Distortion: Changing the size of a file in a non-proportional manner. Also known as "Anamorphic Scaling."

Distribution Amplifier (DA): A device used to replicate an input signal, typically providing 6 outputs, each of which is identical to the input. May also include delay and/or cable equalization capabilities.

Dither: A low level which is added to an analog signal prior to sampling. Typically consists of white noise of one quantizing level peak-to-peak amplitude.

Dolby: A technique developed by Dolby™ Laboratories which improves the signal-to-noise ratio of a recording by a non-linear raise of the volume of specific frequencies in quiet passages before recording, and lowering them to their original levels during playback. The lowering process automatically reduces any noise that was introduced as a result of recording or playback.

DOT 1: A halftone dot (used in color separations). Halftone dots are often confused with pixels but the 2 are not related. Pixels have fixed size by variable density. Halftone dots have fixed density but variable size. This gives the illusion of a continuous tone image when viewed from a distance. There is no fixed relationship between the number of pixels and the number of halftone dots per inch, but a halftone dot can resolve detail smaller than itself (by varying its shape), so for best detail there should be at least twice as many pixels per inch as halftone dots.

DOT 2.: The minimum addressable point in a dot matrix printer. Dot matrix printers build up an image as a mosaic of tiny dots, each of equal density. To express tone levels, these have to be formed into halftone dots.

DOT 3: A pixel in an input scanner or continuous tone output device (e.g., dye-sublimation printer). Scanner resolution is sometimes quoted in DPI (Dots Per Inch) but this can be misleading because here the word "dot" really means "pixel." When referring to a continuous tone scanner, DPI should be changed to PPI (Points Per Inch or Pixels Per Inch) or even to LPI (Lines Per Inch) to avoid confusion.

Dot Pitch: The distance in millimeters between individual dots on a monitor screen. The smaller the dot pitch the better, since it allows for more potential dots to be displayed, giving you better resolution.

Downlink: The communications path from a satellite to its ground station or from a transmitter to a studio.

Downstream Keyer: A keyer that inserts the key after the effects system video output. This enables the key to remain on-air while the background and effects keys are changed behind it.

DPI or DPM: Dots Per Inch (Pixels Per Inch) or Dots Per Millimeter. Can either relate to pixels in an input file or line screen dots (halftone screen) in a pre-press output film. See DOT.

Drop-Frame Time Code: SMPTE time code format that continuously counts 30 frames per second but drops 2 frames from the count every minute except for every tenth minute (drops 108 frames every hour) to maintain synchronization of time code with clock time. This is necessary because the actual frame rate of NTSC video is 29.94 frames per second rather than an even 30 frames. See Non-Drop Frame Time Code.

Dropout: A momentary loss or deterioration of video or audio during playback on a tape machine. Caused by momentary loss of tape contact with the playback head or by flaws in the tape.

Drum Scanner: 1. An optical scanner which mounts the original image on a drum and rotates the drum rapidly while leaving the sensors in place. The scan is created by moving the original image. 2. A high performance input scanner built like a lathe, on which the original is taped to the surface of a spinning drum. A narrow beam of light passes through the drum and enters a lens behind which beam splitters and photo multipliers detect the varying light reflected from the image on the drum and produce digital electrical signals. The lens assembly moves horizontally while the drum is spinning to produce a very fine spiral "raster" or scanning pattern.

DS1: A telephone company format for transmitting information digitally. DS1 has a capacity of 24 voice circuits at a transmission speed of 1.544 megabits per second.

DS3: A telephone company format for transmitting information digitally. DS3 has a capacity of 672 voice circuits at a transmission speed of 44.736 megabits per second.

Dubbing: Transcribing from one recording medium to another.

DVE: Abbreviation of Digital Video Effects. A registered trademark of Nippon Electric Company.

Dynamic Range: The difference between the smallest amount and the largest amount that a system can represent. The dynamic range of an EIM system is the difference between the lightest highlight and the D-Max that the system can scan, manipulate and write.

Edit Code: A tape retrieved code added to original recorded material utilizing a time structure—such as SMPTE time code.

EDL: Abbreviation of Edit Decision List.

E-E Mode: This stands for "electronics to electronics" and is a VTR mode in which the VTR processes the signals it would normally use during recording but does not actually record onto the tape.

EEPROM: Abbreviation of Electrically Erasable Programmable Read Only Memory. A type of memory chip that can hold data even when power is removed. The memory can be erased electronically so new data can be stored.

EFP: Abbreviation of Electronic Field Production, meaning to produce a video production in the field instead of in a studio.

EIA: Electronic Industries Association (formerly RMA or RETMA). The organization which determines recommended audio and video standards in the United States.

EIA Sync: RS-170 sync; the standard waveform for broadcast equipment in the United States.

Encoded: The encoded video signal is formed by starting with an RGB signal from the color television camera. This RGB signal is then processed through an I and Q encoder which converts the RGB into a composite NTSC signal. The encoded signal has all of the elements of the composite video signal: sync, burst, chroma and luminance.

Encoder: A device that superimposes electronic signal information on other electronic signals.

ENG: Abbreviation of Electronic News Gathering, meaning to use a portable video camera and recorder to record news events in the field.

EPROM: Abbreviation of Erasable Programmable Read Only Memory. A type of memory chip that can hold data even when power is removed. The memory can be erased (usually by ultraviolet light exposure) so that new data can be stored.

Equalizer: 1. Equipment designed to compensate for loss and delay frequency effects within a system. 2. A component, or circuit, which allows for the adjustment of a signal across a given band.

External Key: A video key that uses an external key signal (a signal coming from a source outside the device in question) to cut the key hole and a separate fill signal to fill the hole.

Fiber Optic: A transmission designed to transmit signals in the form of pulses of light. Fiber optic cable is noted for its properties of electrical isolation and resistance to electrostatic and electromagnetic interference.

Field: One-half of a television picture. One complete vertical scan of the picture, containing 262.5 lines. 2 fields make up a complete television picture (frame). The lines of Field 1 are vertically interlaced with Field 2 for 525 lines of resolution.

Fill: In video keying, the fill is the video signal that is inserted into the "hole" cut in the background video by a key signal. See Key.

Fill Light: A fill light is used in studio lighting to mask the "mistakes" created by the individual doing the lighting. It is the job of the fill light to cover up and fill the shadow created by the key light.

Film Recorder: A device for converting digital data into film output. Continuous tone recorders produce color photographs, either as transparencies, prints or negatives. Halftone recorders produce film with halftone dots that can be used to make printing plates.

Flat Bed Scanner: An optical scanner that moves the original image and keeps the sensors (usually a CCD array) in place.

Flicker: An annoying picture distortion, mainly related to vertical syncs and video fields display. Some flicker normally exists due to interlacing, more apparent in 50Hz systems (PAL). Flicker shows also when static images are displayed on the screen, i.e., computer generated text transferred to video. Poor digital image treatment, found in low quality system converters (going from PAL to NTSC and vice versa) creates an annoying flicker on the screen. There are several electronic methods to minimize flicker.

F Number: In lenses with adjustable irises, the maximum iris opening is expressed as a ratio (focal length of the lens)/(maximum diameter of aperture). This maximum iris will be engraved on the front ring of the lens.

Focal Length: The distance from the center of the lens to a plane at which point a sharp image of an object viewed at an infinite distance from the camera is produced. The focal length determines the size of the image and the angle of the field of view seen by the camera through the lens. That is the distance from the center of the lens to the pickup device.

Forced Foreground: A feature of some keyers. Uses a mask to force key fill video to appear wherever the mask occurs and completely inhibit background video. Useful for correcting the poor quality key (mixed background and fill) that results when the keying image is poorly differentiated from other images in the key source picture.

Format: In recording of video, C, U-Matic, Betacam, M, Betacam SP, M-II, D1, D2, D3, D-5, Digital Betacam, Beta, VHS, Hi8, 8mm and S-VHS are all current formats.

Frame: 1. The total area of the picture which is scanned while the picture signal is not blanked. 2. A complete TV picture consisting of 2 fields; a total scanning of all 525 lines of the raster area; occurs every 1/30 of a second. (625 lines, 1/25 sec. in Europe and many other countries).

Frame Buffer: Memory used to store a complete frame of video.

Frame Synchronizer: A digital buffer that, by storage and comparison of sync information to a reference, and timed release of video signals, can continuously adjust the signal for any timing errors.

Frequency: The number of complete cycles of a periodic waveform that occur in a given length of time. Usually specified in cycles per second (Hertz).

Frequency Modulation (FM): Modulation of a sine wave or "carrier" by varying its frequency in accordance with amplitude variations of the modulating signal.

Fresnel Lens: A specially constructed lens which produces a soft-edged concentration of light; used as a lens in a spotlight lamp housing.

Front Porch: The blanking signal portion which lies between the end of the active picture information and the leading edge of horizontal sync.

Gain: Any increase or decrease in strength of an electrical signal. Gain is measured in terms of decibels or number of times of magnification.

Gamma Correction: A process used with video and computer graphics images to correct brightness and internal microcontrast within the image. Gamma correction allows a change of ratio between the brightest red component in an image and the weakest red.

Gate: 1. A signal used to trigger the passage of other signals through a circuit. 2. A digital logic device whose output state depends on the states of the logic signals presented to its inputs.

Gamut: The range of voltages allowed for a video signal, or a component of a video signal. Signal voltages outside of the range (i.e., exceeding the gamut) may lead to clipping, crosstalk or other distortions.

General Purpose Interface (GPI): 1. A parallel interconnection scheme that allows remote control of certain functions of a device. One wire per function. 2. May also refer to any non-specific interface between equipment. Usually refers to a serial connection (RS-232 or RS-422 format) between computer modules.

Generations: The number of times a video clip is copied or processed. In analog systems, extensive efforts are made to keep generations to a minimum, since each copy or process adds noise and other artifacts. In digital systems, however, this requirement is no longer necessary, since each copy can potentially be perfect. This enables digital systems to work in quite different ways from analog systems.

Genlock: Genlock is a process of sync generator locking. This is usually performed by introducing a composite video signal from a master source to the subject sync generator. The generator to be locked has circuits to isolate vertical drive, horizontal drive and subcarrier. The process then involves locking the subject sync generator to the master subcarrier, horizontal and vertical drives so that the result is that both sync generators are running at the same frequency and phase.

Ghost: A shadowy or weak image in the received picture, offset either to the right or to the left of the primary image. It is the result of transmission conditions where secondary signals are created and received earlier or later than the primary signal caused by a reflected RF signal.

Gigabyte: Unit of computer memory consisting of about one thousand million bytes (a thousand megabytes). Actual value is 1,073,741,824 bytes.

Gray Scale: A series of tones which range from true black to true white, it is usually expressed in 10 steps.

Grid: A cross hatch of metal pipes for hanging lights in a studio.

Ground Loop: A condition when 2 or more paths to ground exist and a voltage is induced unequally in these paths, causing interference, such as hum, buzz or noise.

H Blanking Width: The width in terms of time occupied by horizontal blanking. The period of time from the end of active video of one line to the beginning of active video of the next line. During this time, the electron beam in a camera or monitor is turned off as it returns or retraces to the other side of the raster to begin a new scan.

HDTV: High Definition Television. The SMPTE in the USA and BTA in Japan have proposed a high definition television product standard: 1125 lines at 60Hz field rate 2:1 interlace; 16:9 aspect ratio; 30MHz RGB and luminance bandwidth; tri-level syncs

Helical Scan: A method of recording video information on a tape, most commonly used in home and professional VCRs. The tape is scanned in a helical way (slanted) rather than horizontally or vertically. The helical scan method allows much more information to be included on a given length of magnetic tape than any other method.

HGA: Hercules Graphics Adaptor.

Hi-Color: An advanced computer graphics format, beyond VGA and Super-VGA, allowing a display of 32,000 colors on the screen at 640 x 480 and 800 x 600 pixels resolution. An even higher number of simultaneous colors displayed on the screen—64,000—is also offered by some card manufacturers. The number of color shades simultaneously displayed on the screen exceeds the color resolution of the human eye, which can resolve about 4000 different color shades.

Hi8: 8mm professional NTSC recording format.

Horizontal Drive also **Horizontal Sync:** This signal is derived by dividing subcarrier by 227.5 and then doing some pulse shaping. The signal is used by monitors and cameras to determine the start of each horizontal line.

Horizontal Resolution: Chrominance and luminance resolution (detail) expressed horizontally across a picture tube. This is usually expressed as a number of black to white transitions or lines that can be differentiated. Limited by the bandwidth of the video signal or equipment.

Horizontal Retrace: At the end of each horizontal line of video, a brief period when the scanning beam returns to the other side of the screen to start a new line.

Horizontal Sync Pulse: The synchronizing pulse at the end of each video line that determines the start of horizontal retrace.

House Sync: Television sync generated within the studio and used as a reference for generating and/or timing other video signals.

H Phase: 1. The horizontal phase relationship of one piece of equipment to another for studio timing purposes. 2. The phase of horizontal sync in relation to subcarrier. See SC/H phase.

Hue (Tint, Phase, Chroma Phase): One of the characteristics that distinguishes one color from another. Hue defines color on the basis of its position in the spectrum, i.e., whether red, blue, green or yellow, etc. Hue is one of the 3 characteristics of television color: See also Saturation and Luminance. In NTSC and PAL video signals, the hue information at any particular point in the picture is conveyed by the corresponding instantaneous phase of the active video subcarrier.

Humbucker: A circuit (often a coil) that introduces a small amount of voltage at power line frequency into the video path to cancel unwanted AC hum.

Hybrid Circuit: A circuit that looks very much like a subminiature printed circuit board and is composed of a mix of thick film and surface mounted components. Hybrids make possible improved performance, extended reliability and economy of space. Use of hybrids permits design of equipment such as entire processing amplifiers (GVG 7510 Series) on single PC modules.

Icon: In a Graphical User Interface (GUI), an on-screen symbol that represents a program file, data file or some other computer entity or function.

IEEE: Institute of Electrical and Electronic Engineers.

IEEE Scale: A waveform monitor scale with the IEEE standards and the recommendations of the TV Broadcasters and Manufacturers for coordination of Video Levels.

Impedance: The total of the resistive and reactive opposition, measured in ohms, that a circuit presents to the flow of alternating current at a given frequency.

Input Scanner: An optical device used to convert drawings or photographs into high resolution digital data. Various types including "array," "drum" and "flying-spot" use different methods to illuminate the image in a pattern of parallel lines or "raster." The reflected or transmitted light is analyzed through red, green and blue filters and digitized into a stream of "pixels." The digital signal can be stored for later processing or sent directly to a film recorder. Input scanners vary in sharpness, color fidelity, speed, cost and ease of operation. High level models allow full control of tone, color and sharpening plus the option to produce either CMYK or RGB signals. Lower-priced models are RGB only and may deliver generally lower quality.

Insert Edit (insert mode): An edit mode in which the time code and control track already existing on the record tape are not replaced during the editing process. The system edits using the pre-recorded control track and time code.

Interface: 1. To connect 2 or more components to each other so the signal from one is supplied to the other(s). Feeding a signal between units that run on different standards. 2. The place where 2 systems or a major and a minor system meet and interact with each other.

Interlaced: Short for interlaced scanning. Also called line interlace. A system of video scanning whereby the odd- and even-numbered lines of a picture are transmitted consecutively as 2 separate interleaved fields.

Intermodulation Distortion (IMD): Distortion that results when 2 or more pure tones produce new tones with frequencies representing the sum and/or difference of the original tones and their harmonics.

Interpolation: In digital video, the creation of new pixels in the image by some method of averaging the values of neighboring pixels. This is necessary when an image is digitally altered, such as when the image is expanded or compressed.

IRE (Institute of Radio Engineers): Units of measurement dividing the area from the bottom of sync to peak white level into 140 equal units. 140 IRE equals 1V p-p. The range of active video is 100 IRE.

Iris: The amount of light transmitted through a lens is controlled by an adjustable diaphragm, or iris, located in the lens barrel. The opening is referred to as the aperture, and the size of the aperture is controlled by rotating the aperture control ring on the lens barrel. The graduations on the lens barrel are expressed in terms of the focal length *f* of the lens divided by the diameter of the aperture at that setting. This ratio is called the *f*-number.

Jitter: Small and rapid variations in a waveform due to mechanical disturbances, changes in the characteristics of components, supply voltages, imperfect synchronizing signals, circuits, etc.

JPEG/MPEG: Standards of storage and retrieval of compressed still and video images, as used in multimedia, video and computer graphics applications. The standards are based on specific hardware and software algorithms.

Kelvin: Also expressed as Kelvins or K, the unit of measurement of the temperature of light. In color recording, light temperature affects the color values of the lights and the scene that they illuminate.

Key: 1. Also called key source or key cut. A signal that can be used to electronically "cut a hole" in a video picture to allow for insertion of other elements such as text or another video image. The key signal is a switching or gating signal for controlling a video mixer which switches or mixes between the background video and the inserted element. 2. The composite effect created by cutting a hole in one image an inserting another image into the hole.

Key Frame: An effect that has been stored in memory, similar to a snapshot photograph. Individual key frames can be strung together to create an overall key frame effect similar to animation.

Ramp: A video test signal that graduates from low luminance to high luminance used to measure luminance linearity.

Raster: 1. Pixel based image information file, in which the image is expressed by a very fine grid of numerical brightness values. Each grid cell, or pixel, is stored as a set of numbers for CMYK, RGB, or intensity, hue and saturation values. Content of image, tint and hue are recorded pixel by pixel in order of location.

Read: The function of copying a file from tape to disc or from disc or tape to RAM.

Read Before Write: A feature of some videotape recorders that plays back the video or audio signal off of tape before it reaches the record heads, sends the signal to an external device for modification, and then applies the modified signal to the record heads so that it can be re-recorded onto the tape in its original position.

Reflected Light: The scene brightness or the light being reflected from a scene. Usually it represents 5 to 95% of the incident light, and it is expressed in footlamberts.

Registration: An adjustment associated with color sets and projection TVs to ensure that the electron beams of the 3 primary colors of the phosphor screen are hitting the proper color dots/stripes; also, a similar adjustment of the tubes in color cameras.

Repeater: 1. A receiver/transmitter that receives a signal from another transmitter and relays (retransmits) it to another receiver or a receiver/transmitter. 2. In fiber optics, a device that converts a received optical signal to its electrical equivalent, reconstructs the source signal format, amplifies and reconverts it to an optical output signal. The purpose is to restore the light amplitude, compensating for normal loss in fiber.

Resolution: 1. A measure of the ability of a camera or television system to reproduce detail. That is the number of picture elements that can be reproduced with good definition. It is a factor of the pickup device or the TV CRT characteristics and the video signal bandwidth. 2. Generally called horizontal resolution. It can be evaluated by establishing the limit to which lines can be distinguished on a test pattern. A larger resolution value means a broader frequency band of the video signal. 3. A measure of the greatest amount of detail that can be seen, or resolved, in an image. Often incorrectly expressed as a number of pixels on a given line. More correct is the bandwidth.

`etrace: The return of the electron beam in a CRT to the starting point after :canning. During retrace, the beam is typically turned off. All of the sync information is placed in this "invisible" portion of the video signal. May refer to retrace after each horizontal line or after each vertical scan (field).

RGB, RGB Format, RGB System: Red, Green and Blue: The basic parallel component set in which a signal is used for each primary color; or the related equipment or interconnect formats or standards. The same signals may also be called "GBR" as a reminder of the mechanical sequence of connections in the SMPTE interconnect standard.

RIP (Raster Image Processor): Used to convert vector images to raster images in computers using both kinds of image files.

Rise Time: The time taken for a signal to make a transition from one state to another; usually measured between the 10% and 90% completion points of the transition. Shorter, or "faster" rise times require more bandwidth in a transmission channel.

RMS: Abbreviation of Root Mean Square. A measure of effective (as opposed to peak) voltage of an AC waveform. For a sine wave it is 0.707 times the peak voltage. For any periodic waveform, it is the square root of the average of the squares of the values through one cycle.

ROM: Abbreviation of Read Only Memory. A memory device that is programmed only once with a permanent program or data that cannot be erased.

Routing Switcher: An electronic device that routes a user-supplied signal (audio, video, etc.) from any input to any user-selected output. Inputs are called sources. Outputs are called destinations.

RP-125: A SMPTE parallel component digital video standard.

RS-170A: A document prepared by the Electronics Industries Association describing recommended practices for NTSC color television signals in the United States.

RS-232: A standard, single-ended (unbalanced) interconnection scheme for serial data communications.

RS-250B: In telecommunications, a transmission specification for NTSC video and audio.

RS-422: A standard, balanced interconnection scheme for serial data com-'nications.

ɯafe Title Area: 80% of the TV screen, from the center of the screen; that area of the display screen (and therefore of the camera scanning area) which will reproduce legible title credits no matter how it is adjusted.

Satellite Downlink: The communications path from a satellite to its ground station.

Satellite Uplink: The communications path from a ground station to its satellite.

Saturation (Chroma, Chroma Gain, Color): 1. The intensity of the colors in the active picture. The voltage levels of the colors. The degree by which the eye perceives a color as departing from a gray or white scale of the same brightness. A 100% saturated color does not contain any white; adding white reduces saturation. In NTSC and PAL video signals, the color saturation at any particular instant in the picture is conveyed by the corresponding instantaneous amplitude of the active video subcarrier. 2. The point on the operational curve of an amplifier at which an increase in input amplitude will no longer result in an increase in amplitude at the output.

Scale: A change in the dimensional size of a file without changing the data that make up the file.

Scanning: The rapid movement of the electron beam in a pickup device of a camera or in the CRT of a television receiver. It is formatted in a line-for-line manner across the photo sensitive surface which produces or reproduces the video picture. When referred to a video surveillance field, it is the panning or the horizontal camera motion.

SC/H Phase (Subcarrier to Horizontal Phase): In NTSC video, the phase relationship of the subcarrier to the leading edge of horizontal sync. SC/H phase is correct when the zero crossing of subcarrier is aligned with the 50% point of the leading edge of sync. In PAL video, the SC/H phase is defined as the phase of the E_U component of the color burst extrapolated to the half amplitude point of the leading edge of synchronizing pulse of line 1 of field 1.

Scoop: A large bowl-shaped unit—often made of aluminum—into which a lighting unit is placed so that it will reflect light over a wide area.

SECAM: Abbreviation of Sequential Couleur Avec Memoire (sequential color with memory). A color television system with 625 lines per frame and 50 fields per second developed by France and the U.S.S.R. Color difference information is transmitted sequentially on alternate lines as an FM signal.

SEG (Special Effect Generator): The SEG is used in multi-camera production and editing to change from one camera (or VCR) signal to another. Many different changes or "wipes" are possible. For this to work properly, all connected equipment must be driven by the same sync signal (often provided by a sync generator built into the SEG itself).

Sepia: A process used in photography to generate a brownish tone in pictures, providing an "antique" appearance. The same idea has been electronically adapted in video special effects generation. A color picture or a black and white picture can be colored in sepia.

Sequential Assembly: In video editing, a sequential method of auto assembly. The computerized editing system records all edits listed in the edit decision list in order from first to last, requesting source tapes as they are needed. Also called A-mode assembly. See also Auto Assembly.

Serial: Time-sequential transmission of data along a single wire. Analogous to a railroad train, where each car (data bit) follows the other in single file.

Serial Digital: Digital information that is transmitted in serial form. Often used informally to refer to serial digital television signals.

Serial Interface: A digital communications interface in which data is transmitted and received sequentially along a single wire or pair of wires. Common serial interface standards are RS-232 and RS-422.

Serializer: A device that converts parallel digital information to serial.

Serial Port: A computer I/O (Input/Output) port through which the computer communicates with the external world. The standard serial port is RS-232 based and allows bi-directional communication on a relatively simple wire connection as data flows serially.

Setup (Black Reference, Black Level): 1. The specified base of the active picture signal which is at reference black level. Called setup because it is placed 7.5 IRE units above blanking (zero IRE) in NTSC video. 2. The basic operating configuration of a system.

SGO: Second Generation Original or Second Original. Usually refers to a film output made from a computer data file that represents image quality as good as the original camera transparency or negative.

Signal-to-Noise Ratio—S/N: A S/N ratio can be given for the luminance signal, chrominance signal and audio signal. The S/N ratio is the ratio of noise to actual total signal, and it shows how much higher the signal level is than the level of noise. It is expressed in decibels (dB), and the bigger the value is, the more crisp and clear the picture and sound will be during playback.

Single-Mode Fiber: An optical glass fiber that consists of a core of very small diameter (usually 2-10 microns) and a cladding approximately 20 times the thickness of the core. Such fibers are normally used only with laser sources because of their very small acceptance cone. Since the cone diameter approaches the wavelength of the source, only a single mode is propagated.

Skewing: Due to loss or distortion of equalizing pulses and serrations, found mainly in multi-generation video tapes, the upper third of the video picture is "flagging" sidewise or skewing. To overcome the problem for consumer equipment, the television is equipped with an "AV channel," that when selected for VCR viewing purposes, certain time constants in the sync regeneration circuits are changed, allowing viewing without skewing. In the professional studio this problem is solved using a TBC.

S-MAC: A MAC standard proposed for studio intraconnection by the SMPTE working group on Component Analog Video Standards: The S-MAC system uses time compression and time domain multiplexing techniques to convey Y, C_r, C_b video signals (a version of Y, R-Y, B-Y).

SMPTE: Society of Motion Picture and Television Engineers.

SMPTE Time Code: Time code that conforms to SMPTE standards. It consists of an 8-digit number specifying hours: minutes: seconds: frames. Each number identifies one frame on a videotape. SMPTE time code may be of either the drop-frame or non-drop frame type. In GVG editors, the SMPTE time code mode enables the editor to read either drop-frame or non-drop frame code from tape and perform calculations for either type (also called mixed time code).

Snow: 1. Random noise on the display screen, often resulting from dirty heads. 2. TV signal breakup caused by weak video reception.

Soft Edge Masking: A process used in image processing to increase the apparent sharpness of an image. The computer analyzes the pixels and makes soft edges of objects into sharp edges.

SONET: Abbreviation of Synchronous Optical Network. A telecommunications standard.

Spectral Bandwidth: In telecommunications, the spectral bandwidth for single peak devices is the difference between the wavelengths at which the radiant intensity is 50% (or 3dB) down from the maximum value.

Split Edit: An edit in which the audio in-edit point is different from the corresponding video in-edit point.

Split Screen: A special effect utilizing 2 or more cameras so that 2 or more scenes are visible simultaneously on each part of the screen.

Staircase: A pattern generated by the NTSC generator, consisting of equal width luminance steps of 0, +20, +40, +60, +80, and +100 IEEE units and a constant amplitude chroma signal at color burst phase. Chroma amplitude is selectable at 20 IEEE units (low stairs) or 40 IEEE units (high stairs). The staircase pattern is useful for checking linearity of luminance and chroma gain, differential gain and differential phase.

Standard, Interconnect Standard: The specific signal configuration, reference pulses, voltage levels, etc. which describe the input/output requirements for a particular type of equipment. Some standards have been established by professional groups or government bodies (such as SMPTE or EBU). Others are determined by equipment vendors and/or users.

Stripe Filter: A chrominance tube system in which the target area of the tube is divided into sequential stripes for RBG and Y, and can therefore derive a color signal by using only one pickup tube.

Subcarrier: Also SC, 3.58, 3.58CW—This is the basic signal in all NTSC sync signals. It is a continuous sine wave, usually generated and distributed at 2 volts in amplitude, and having a frequency of 3.579545MHz. Subcarrier is usually divided down from a primary crystal running at 14.318180MHz, and that divided by 4 is 3.579545. All other synchronizing signals are directly divided down from subcarrier.

Subcarrier Phase Shifter: Special circuitry designed to control the phase relationships of the 2 portions of the encoded color signal so they maintain their correct relationship during recording, transmission and reproduction.

S-Video: Superior Video, a widely accepted set of Y/C signals used to connect video equipment, providing a higher quality signal free of the cross luminance/color problems associated with composite video signals.

Switcher: Term often used to describe a special effects generator, a unit which allows the operator to switch between video camera signals. Switchers are often used in industrial applications to switch between video cameras monitoring certain areas for display on one monitor; these kinds of switchers do not have sync generators.

Sync: The portion of an encoded video signal that occurs during blanking and is used to synchronize the operation of cameras, monitors and other equipment. Horizontal sync occurs within the blanking period in each horizontal scanning line and vertical sync occurs within the vertical blanking period.

Sync Generator (Sync Pulse Generator, SPG): Device that generates synchronizing pulses needed by video source equipment to provide proper equipment or studio timing. Pulses typically produced by a sync generator include subcarrier, burst flag, sync, blanking, H & V drives, color frame identification and color black.

T1: In telecommunications, the paired cable used to transport DSI service.

Tally: 1. A lamp which lights to indicate that the associated video source is in use. Typical locations of tally lamps are on the front of video cameras and in the crosspoint push-buttons of video switchers. 2. The acknowledgement returned to the control panel or terminal that an operation has been executed.

TBC (Time Base Corrector): This piece of equipment corrects the timing irregularities that occur during VCR playback. Time base correction is not necessary for direct playback from a VCR to a TV set.

Tearing: A lateral displacement of the video lines due to sync instability. Visually it appears as though parts of the images have been torn away.

Telecine: Telecine is a device mainly designed to convert film to video. The movie film in advanced telecine machines is sampled digitally and converted to video, frame after frame, in real time. One of the most popular digital systems used in professional telecine machines is called "flying-spot," allowing almost a transparent conversion to video. The main problem encountered in film-to-video conversion is the frame rate. Movie film has a frame rate of 18 or 24 frames per second, and neither the PAL nor NTSC systems has a similar frame rate. In order to have a good conversion, interpolation and other techniques are used in the telecine device.

Teleconferencing: Electronically-linked meeting conducted among groups in separate geographic locations.

Teleprompting: Text shown on a television monitor to assist a performer or speaker.

Terminate, Termination: To complete a circuit by connecting a resistive load to it. A video termination is typically a male BNC connector which contains a 75 ohm resistive load. When there are looping inputs, any unused looping input must be terminated in 75 ohms to ensure proper signal levels and to minimize reflections.

Test Pattern: Optical guide for TV camera reference alignment.

Texture Mapping: The ability of a digital picture manipulator to create textured surfaces that can be applied to shapes.

TFT Screen: TFT stands for Thin-Film-Transistor. This technology is used mainly for manufacturing flat computer and video screens that are superior to the classic LCD screens. Color quality, fast response time and resolution are excellent for video.

Time Base Error: Horizontal rate flutter of a video signal caused by tape stretch and inherent imperfections in the tape transport mechanism of a videotape recorder.

Time Base Stability: The maintenance of the scanning process to very close tolerances.

Time Code Editing: By recording a sequential time code along with the video and audio material, you can obtain a more precise reference for editing. Each frame has its own number or code which tells the time in hours, minutes, and seconds, and includes a frame number. The world standard code is called SMPTE (Society of Motion Picture and Television Engineers) and has also been adopted by the IEC (International Electrotechnical Commission). Time codes permit very fast and accurate editing. Automatic editing is possible under computer control.

Timeline: An effects control feature that enables the operator of a switcher or digital picture manipulator to pre-program a series of timed events, such as auto transitions, E-MEM recalls and GPI triggers, and then replay them.

Title Generator: Commonly a black and white camera is used to shoot titles which are electronically superimposed on the video picture while shooting or during editing. Title color can be selected and changed independently. More sophisticated equipment generates characters directly.

T-Pulse to Bar: A term relating to frequency response of video equipment. A video signal containing equal amplitude T-pulse and bar portions is passed through the equipment and the relative amplitudes of the T-pulse and bar are measured at the output. A loss of response is indicated when one portion of the signal is lower in amplitude than the other.

Tracking: The angle and speed at which the tape passes the video heads.

Transcoder: A device that converts one form of encoded video to another, e.g., to convert NTSC video to PAL. Sometimes mistakenly used to mean translator.

Transducer: A device that converts one form of energy into another. For example, in fiber optics, a device that converts light signals into electrical signals.

Translator: A device used to convert one component set to another, e.g., to convert Y, R-Y, B-Y signals to RGB signals.

Triaxial: This is a connector comprised of three concentric conductors, an inner conductor, intermediate conductor and outer conductor, separated by dielectrics.

TTL (Transistor-Transistor Logic): A term used in digital electronics mainly to describe the ability of a device or circuit to be connected directly to the input or output of digital equipment. Such compatibility eliminates the need for interfacing circuitry. TTL signals are usually limited to 2 states, low and high, and are thus much more limited than analog signals.

Twinax: This is a connector which has 2 insulated inner contacts (male and female) surrounded by a common ground.

Twisted Pair: A cable composed of 2 small insulated conductors twisted together. Since both wires have nearly equal exposure to any interference, the differential noise is slight.

Unbalanced: Frequently, a circuit having one side grounded. A circuit, the 2 sides of which are electrically different.

Underscan: Decreases raster size H and V so that all 4 edges of the picture are visible on the monitor. Allows viewing of skew and tracking which would not be visible in normal (overscanned) mode. Also helpful when aligning test charts to be certain they touch all four corners of the raster. Likewise, when checking the alignment of multiplexer images from a film chain, underscan allows proper framing of the projected image going into the video camera.

Upstream: 1. Placed ahead of other devices in a video signal path. 2. Describes the location of keyers in a mix/effects level or in the overall switcher architecture. 3. Relates to the priority of the video signals as they are combined through the video production switcher.

Valid Signal: A video signal which will remain legal when transcoded to any other format. A valid signal is always legal, but a legal signal is not necessarily valid. Signals which are not valid will be processed without problems in their current format, but problems may be encountered if the signal is transcoded to a new format.

Vector Image: An image system that uses basic geometric shapes like rectangles, lines, circles, ellipses and polygons to create a graphic image. The vector image usually contains very little data, like the starting point (pixel) of the object, what kind of object it is, its size and color. When the image is rasterized, the vector information is converted into a bitmap using a RIP (Raster Image Processor). Rescaling can be performed with greater accuracy than with raster data. Also called "object oriented."

Vectorscope: Round (green) oscilloscope to align amplitude and phase of the 3 TV color signals (RGB).

Vertical Interval: The portion of the video signal that occurs between the end of one field and the beginning of the next. During this time, the electron beams in the cameras and monitors are turned off (invisible) so that they can return from the bottom of the screen to the top to begin another scan.

Vertical Interval Switching: When one video signal is replaced by another, the switching process causes a random interruption in the first video signal which may be in the midst of a frame) and a random entrance into the second video signal (also in the middle of a frame). The result is a jump in the picture when the edited tape is played. This situation is amplified when the tape is copied, and the disturbance on playback is much more serious. To avoid this problem, switching is performed at a very specific point—during the vertical blanking retrace period—which is also known as the vertical interval. This allows complete replacement of a whole frame by a second whole frame and the switching process is very smooth.

Vertical Resolution: Chrominance and luminance detail expressed vertically in the picture tube. Limited by the number of scan lines.

Vertical Retrace: The return of the electron beam to the top of a television picture tube screen or a camera pickup device target at the completion of the field scan.

Vertical Sync Pulse: A portion of the vertical blanking interval which is made up of blanking level and 6 pulses (92% duty cycle at -40 IEEE units) at twice the horizontal sync pulse repetition rate. Synchronizes vertical scan of television receiver to composite video signal. Starts each frame at same vertical position (sequential fields are offset 1/2 line to achieve interlaced scan).

VGA: Video Graphics Array.

Video: Pertaining to picture signals in a television system.

Video Bandwidth: The highest signal frequency that a specific video signal can reach. The higher the video bandwidth, the better the quality of the picture. A video recorder that can produce a very broad video bandwidth generates a very detailed, high quality picture on the screen. Video bandwidths used in studio work vary between 3 and 12MHz.

Video Distribution Amplifier: A special amplifier for strengthening the video signal so that it can be supplied to a number of video monitors at the same time.

Video Gain (White Level, White Bar, Reference White): The range of light-to-dark values of the image which are proportional to the voltage difference between the black and white voltage levels of the video signal. Expressed on the waveform monitor by the voltage level of the whitest whites in the active picture signal. Video gain is related to the contrast of the video image.

Video Printer: The video printer is a special device for grabbing a video frame to create a hard copy of that frame on photographic-like paper. Most video printers allow the capture of one video frame to create, in several seconds to a few minutes, a hard copy of the video frame. Video printers are useful in several industrial, as well as in medical and military applications. The quality of the print is limited by the quality of the picture on the screen, but for most applications it suffices.

Video Tape Recorder: VTR; an electro-mechanical device capable of recording, storing and reproducing an electronic signal which contains audio, video and control information.

Video Wall: A video wall is large screen made up of several monitors which are placed close to one another, so when viewed from a distance, form a large video screen or "wall." In order to break down the video image into several segments, each filling up its own monitor, a special unit is needed. The video wall generating unit is a digitally-based processor which converts the analog video signal to digital. It rescans and resamples it and generates several individual analog video outputs, each driving one monitor. The unit is relatively expensive and serves a similar purpose as a wide screen projector, mainly for large audiences.

Video Waveform: The pictorial display on a special oscilloscope of the various components of the video signal, used to check the integrity of the signal and signal components.

Videocassette: A self-contained video module played on a specially designed video tape recorder; similar in design to an audio cassette; houses 2 reels—supply and take-up with the tape running between them but connected to both.

VITC: (Vertical Interval Time Code): This is the same information as the SMPTE time code. It is superimposed onto the vertical blanking interval, so that the correct time code can be read even when a helical scanning VCR is in the Pause or Slow mode.

VITS (Vertical Interval Test Signal): A signal that may be included during the vertical blanking interval to permit on-the-air testing of video circuit functions and adjustments.

VTR: Video Tape Recorder. The term "VTR" includes reel-to-reel and cassette type.

Watt: A measure of electrical power. The power expended when one ampere of direct current flows through a resistance of one ohm. The unit of electric power required to do work at the rate of one joule per second. Calculated by multiplying volts times amperes.

Waveform Monitor: Oscilloscope used to display the video waveform.

White Balance: An electronic process used in video cameras to retain true colors. White balancing is performed prior to the recording of a specific scene. The camera is pointed at a white object (a wall, for example) and controls on the camera are adjusted until a hairline in the viewfinder is brought to a particular point. This ensures that the tints in the video tape will be natural. Unnatural colors are the result of incorrect white balance. To correct this situation a chroma corrector is used to restore the white balance and the normal tints.

Wipe: Term used to describe the SEG effect of replacing a portion of video signal A with video signal B; also to erase a tape.

Wow and Flutter: Wow refers to low frequency variations in pitch while flutter refers to high frequency variations in pitch caused by variations in the tape-to-head speed of a tape machine.

Write: A function of copying a file from disc to tape. Also sometimes used to describe the transfer of information from the internal computer memory to a disc.

Y/C: A set of video signals that contain a separate Y, which is luminance, and C, which is chroma. Usually the chroma is at 3.58MHz, as in the S-Video signal, but it can also be at 688kHz in the 3/4" dub format.

Y, C1, C2: A generalized set of CAV signals: Y is the luminance signal, C1 is the first color difference signal and C2 is the second color difference signal.

Y, I, Q: The set of CAV signals specified for the NTSC system: Y is the luminance signal, I is the first color difference signal and Q is the second color difference signal.

Y, P_b, P_r: A version of Y, R-Y, B-Y specified for the SMPTE analog component standard.

Y, R-Y, B-Y: The general set of CAV signals used in the PAL system as well as for some encoder and most decoder applications in NTSC systems; Y is the luminance signal, R-Y is the first color difference signal and B-Y is the second color difference signal.

Y, U, V: Luminance and color difference components for PAL systems; Y, B-Y, R-Y with new names; the derivation from RGB is identical.

Z Axis: The Z axis is in the third dimension, perpendicular to the X and Y axes and indicates depth. See axis.

Zero Dispersion Point: In telecommunications, the wavelength where material dispersion is minimal. With standard fiber optic cable, that wavelength is 1310 nanometers.

Zero Suppression: In telecommunications, techniques that limit the number of consecutive data 0's that may be transmitted. For DS1 without B8ZS, 15 data 0's is the maximum allowed.

Zoom Ratio: A mathematical expression of the 2 extremes of focal length available on a particular zoom lens.